DEFENSE INFORMATION SYSTEMS AGENCY
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MEMORANDUM FOR DISTRIBUTION   **31 Mar 11**

SUBJECT: Special Interoperability Test Certification of Cisco Unified Communications Manager
           Version 8.0(2) with Internetwork Operating System (IOS) Software Release 15.1(1)T

References: (a) DoD Directive 4630.05, "Interoperability and Supportability of Information
                  Technology (IT) and National Security Systems (NSS)," 5 May 2004
             (b) CJCSI 6212.01E, "Interoperability and Supportability of Information
                  Technology and National Security Systems," 15 December 2008
             (c) through (g), see Enclosure 1

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint
Interoperability Test Command (JITC), as the responsible organization for interoperability test
certification.

2. The Cisco Unified Communications Manager Version 8.0(2) with IOS Software Release
15.1(1)T is hereinafter referred to as the system under test (SUT). The SUT meets all of its
critical interoperability requirements and is certified for joint use within the Defense Information
System Network (DISN) for the following switch types: Private Branch Exchange (PBX) 1 and
PBX 2. The SUT meets the Voice over Internet Protocol (VoIP) critical interoperability
requirements with any certified Assured Services Local Area Network (ASLAN) or ASLAN
components on the Unified Capabilities (UC) Approved Products List (APL). The identified test
discrepancies shown in the Certification Testing Summary (Enclosure 2) have an overall minor
operational impact. No other configurations, features, or functions, except those cited within this
report, are certified by the JITC. This certification expires upon changes that could affect
interoperability, but no later than three years from the date of Defense Information Assurance
(IA)/Security Accreditation Working Group (DSAWG) accreditation.

3. This finding is based on interoperability testing conducted by JITC, DISA adjudication of
open test discrepancy reports, review of the vendor's Letters of Compliance (LoC), and DSAWG
accreditation. Interoperability testing of the SUT was conducted at JITC's Global Information
Grid Network Test Facility at Fort Huachuca, Arizona, from 28 June through 13 August 2010.
DISA adjudication of outstanding test discrepancy reports was completed on 25 August 2010.
Review of the vendor's LoC was completed on 8 December 2010. The Field Security Office
(FSO) granted accreditation on 31 March 2011 based on the security testing completed by DISA-
led IA test teams and published in a separate report, Reference (c). Enclosure 2 documents the
test results and describes the tested network and system configurations.

JITC Memo, JTE, Special Interoperability Test Certification of Cisco Unified Communications Manager Version 8.0(2) with Internetwork Operating System (IOS) Software Release 15.1(1)T

4. The interoperability test summary of the SUT is indicated in Table 1. The PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 2. This interoperability test status is based on the SUT's ability to meet:

    a.   Defense Switched Network (DSN) services for Network and Applications specified in Reference (d).

    b.   PBX 1 interface and signaling requirements for trunks/lines specified in References (e) and (f) verified through JITC testing and/or vendor submission of LoC.

    c.   PBX 1 CRs/FRs specified in References (e) and (f) verified through JITC testing and/or vendor submission of LoC.

    d.   The overall system interoperability performance derived from test procedures listed in Reference (g).

    e.   The Internet Protocol version 6 (IPv6) requirements were met through testing and review of the vendor LoC.

Table 1. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	The SUT T1 CAS interface did not meet all critical CRs and FRs. The SUT T1 CAS interface is not certified by JITC and is not required for a PBX 1.[1]
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	Although this interface is offered by the SUT, it was not tested. The SUT E1 CAS interface is not certified by JITC and is not required for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI2 interface does not support NFAS.[2]
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Met all critical CRs and FRs.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog Loop Start (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT gateway analog interface does not provide a ping ring when the phone is configured with the Call Forward Variable feature.[3]
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 1.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 1.
VoIP (Ethernet IEEE 802.3u)	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The Cisco IP phones do not support Call Waiting.[4] The Cisco CP-7940G and CP-7960G are legacy end instruments that did not meet dual stack IPv6 requirements.[5]
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exception: Full compliance of DSN Common Call Features was not met.[3]
Attendant	No	Not Tested	This feature is not supported by the SUT and is not required for a PBX 1.
Public Safety	Yes	Certified	The SUT met all critical CRs and FRs for Basic 911.[6]
Conferencing	No	Not Tested (See note 7.)	The SUT does not support conferencing. However, the SUT can support Meet-Me Conferencing with the optional Cisco MeetingPlace® Express. The SUT does not support Preset Conferencing or Progressive Conferencing and these features are not required for a PBX 1.
Nailed-up Connections	No	Not Tested	This feature is not supported by the SUT and is not required for a PBX 1.
DSN Hotline Services	No	Not Tested	This feature is not supported by the SUT and is not required for a PBX 1.

Table 1. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities (continued)			
Features and Capabilities	**Critical**	**Status**	**Remarks**
MLPP	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support a global diversion number.[8] The SUT does not support the Loss of Command and Control announcement.[9]
Call Processing	Yes	Certified	Met all critical CRs and FRs.
ISDN Services	Yes	Certified	Met all critical CRs and FRs on the T1 PRI interface with the following minor exception: The SUT T1 ISDN PRI NI2 interface does not support NFAS.[2]
Synchronization	Yes	Certified	Met all critical CRs and FRs.
Reliability	Yes	Certified	Met all critical CRs and FRs.
Security	Yes	Certified	See note 10.
VoIP System	No	Certified	The SUT is certified for VoIP with any certified ASLAN or ASLAN components posted on the UC APL. The SUT also met IPv4/IPv6 requirements. (See notes 5 and 11.)
VoIP Softphone	No	Certified	Met all critical CRs and FRs.
Network Gateways				

Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	The SUT T1 CAS interface did not meet all critical CRs and FRs. The SUT T1 CAS interface is not certified by JITC and is not required for a PBX 1.[1]
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	Although this interface is offered by the SUT, it was not tested. The SUT E1 CAS interface is not certified by JITC and is not required for a PBX 1.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI/2 interface does not support NFAS.[2]
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	2-Wire Analog Ground Start (GR-506-CORE)	No	Certified	Met all critical CRs and FRs.[12]

NOTES:
1   The SUT T1 CAS interface does not recognize Remove from Service (Busy Out) or Restore to Service (Make Idle) condition from the distant end switch. These are critical requirements for this interface; therefore, this interface is not certified by JITC.
2   The SUT does not support NFAS on their T1 ISDN PRI NI2. DISA previously adjudicated this anomaly as having a minor operational impact and stated the intent to change this from required to conditional for a PBX 1.
3   A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.
4   All of the features on the VoIP phones were tested using multiple line appearances. Although the SUT does not support Precedence Call Waiting on their VoIP phones, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call.
5   The SUT met all IPv4 and IPv6 requirements through testing and LoC with the following exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in Enclosure 2 and a dual stack call control agent in accordance with Reference (g). This was adjudicated by DISA as having a minor operational impact.
6   The SUT only supports emergency basic 911 service. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. These public safety features are not required for a PBX 1.
7   The SUT does not support conferencing. However, the SUT can support Meet-Me Conferencing through the use of an optional adjunct conferencing system called the Cisco MeetingPlace® Express, which is covered under a separate certification.
8   The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified Communications Manager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.
9   The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. DISA previously adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1.
10 Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (c).

Table 1. SUT Interoperability Test Summary (continued)

NOTES continued:

11   The SUT met all IPv4 and IPv6 requirements with the following discrepancies noted with the SUT, which were adjudicated by DISA as having a minor operational impact:

  a. The VoIP SUT session control components and end instruments can only have the signaling service Traffic Class configured for 21 different DSCP values and not the full range required of 0-63.
  b. The MCS 7835 and the MCS 7825 call managers OAM traffic is tagged at zero and is not configurable.
  c. The 2851 and 3845 gateways are tagging IPv4 RTCP traffic at zero and it is not configurable.
  d. When the CP-7940G and CP-7960G phones are powered up, some of the UDP/TFTP traffic has a DSCP value of 4 and 802.1Q value of 5 and cannot be changed.
  e. The SUT management workstation provided during testing did not assign DSCP values for OAM IP traffic.
  f. The IP phones are incorrectly tagging IPv6 TCP traffic during power up.
  g. The Soft Client is incorrectly tagging all traffic during power up.
  h. The 802.1Q CoS tag values are not independently configurable from the DSCP values.
  i. End Instruments do not support the manual configuration of the IPv6 default gateway.
  j. Communications Managers are incorrectly tagging UDP/TFTP traffic to the end instrument during end instrument power up.

12   This interface requirement was met by the vendor's LoC.

LEGEND:

802.1Q	Standards for Local and Metropolitan Area Networks: Virtual Bridged Local Area Networks	JITC	Joint Interoperability Test Command
		LoC	Letters of Compliance
		LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	Mbps	Megabits per second
ANSI	American National Standards Institute	MCS	Media Convergence Servers
APL	Approved Products List	MFR1	Multi-Frequency Recommendation 1
ASLAN	Assured Services Local Area Network	MLPP	Multi-Level Precedence and Preemption
BRI	Basic Rate Interface	NI 1/2	National ISDN Standard 1 or 2
C2	Command and Control	NI2	National ISDN Standard 2
CAS	Channel Associated Signaling	NFAS	Non Facility Associated Signaling
CoS	Class of Service	OAM	Operational Administration and Maintenance
CP	Cisco Phone	PBX 1	Private Branch Exchange 1
CRs	Capability Requirements	PRI	Primary Rate Interface
DISA	Defense Information Systems Agency	PSTN	Public Switched Telephone Network
DP	Dial Pulse	Q.931	Signaling Standard for ISDN
DSCP	Differentiated Services Code Point	Q.955.3	ISDN Signaling standard for E1 MLPP
DSN	Defense Switched Network	RTCP	RTP Control Protocol
DSS1	Digital Subscriber Signaling 1	RTP	Real-time Transport Protocol
DTMF	Dual Tone Multi-Frequency	SS7	Signaling System 7
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test
FRs	Feature Requirements	T1	Digital Transmission Link Level 1 (1.544 Mbps)
GR	Generic Requirement	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
GR-506-CORE	LSSGR: Signaling for Analog Interfaces		
IEEE	Institute of Electrical and Electronics Engineers	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IP	Internet Protocol	TCP	Transmission Control Protocol
IPv4	Internet Protocol version 4	TFTP	Trivial File Transfer Protocol
IPv6	Internet Protocol version 6	UC	Unified Capabilities
ISDN	Integrated Services Digital Network	UCR	Unified Capabilities Requirements
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	UDP	User Datagram Protocol
		VoIP	Voice over Internet Protocol

Table 2. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	**Critical**		**Requirements Required or Conditional**	**References**
T1 CAS (MFR1, DTMF, DP)	No	Trunking	• PBX Line (C) • Direct Inward Dialing (C) • National ISDN 1/2 Primary Access (R) • ISDN ANSI MLPP Service Capability (R) • ITU-T ISDN Primary Access (Europe only) (C) • ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C) • Normal Wink Start Operations (R) • Glare Operation (R) • Abnormal Wink Start (R) • Glare Resolution (R) • Call for Service Timing (R) • Guard Timing (R) • Satellite Timing (R) • Disconnect Control (R) • Reselect and Retrial (R) • Off-Hook Supervision Transition (R) • Dial-Pulse Signals (R) • DTMF Signaling (R) • Standard Digit Format for Precedence (C) • MFR1 2/6 Signaling (C) • Alerting Signals and Tones (R) • DSN ISDN User-to-Network Signaling (R) • Application (R) • Physical Layer (R) • Data Link Layer (R) • Data Link Connection (R) • Peer-to-Peer Procedures of Data-Link Layer (R) • Layer 3 DSN User-to-Network Signaling (R) • DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R) • Sequence of Messages for DSN Circuit-Switched Calls (R) • Message Functional Definition and Content (R) • General Message Format and Information Elements Coding (R) • Supplementary Services (C) • PCM-24 Digital Trunk Interface (R) • Interface Characteristics (R) • Supervisory Channel Associated Signaling (R) • Clear Channel Capability (R) • Alarm and Restoral Requirements (R) • PCM-30 Digital Trunk Interface (Europe only) (R) • Interoperation of PCM-24 and PCM-30 (R) • Analog Trunk Interface (C) • Integrated Digital Loop Carrier (C) • Trunk Group-Remove from Service (R) • Trunk Group-Restore to Service (R)	• UCR Section 5.2.1.3.1 • UCR Section 5.2.1.3.2 • UCR Section 5.2.1.3.4.1 • UCR Section 5.2.1.3.4.1.1 • UCR Section 5.2.1.3.4.2 • UCR Section 5.2.1.3.4.2.1 • UCR Section 5.2.4.3.3.1.1 • UCR Section 5.2.4.3.3.1.2 • UCR Section 5.2.4.3.3.2.1 • UCR Section 5..2.4.3.3.2.2 • UCR Section 5.2.4.3.5 • UCR Section 5.2.4.3.6 • UCR Section 5.2.3.4.7 • UCR Section 5.2.3.4.8 • UCR Section 5..2.3.4.9 • UCR Section 5.2.3.4.10 • UCR Section 5..2.4.4.1 • UCR Section 5.2.4.4.2 • UCR Section 5..2.4.4.2.1 • UCR Section 5.2.4.4.3 • UCR Section 5..2.4.5.1 • UCR Section 5.2.4.7.1.4.2 • UCR Section 5..2.4.7.1.1 • UCR Section 5.2.4.7.1.2 • UCR Section 5.2.4.7.1.3 • UCR Section 5..2.4.7.1.3.1 • UCR Section 5.2.4.7.1.3.2 • UCR Section 5.2.4.7.1.4 • UCR Section 5.2.4.7.1.4.2 • UCR Section 5.2.4.7.1.4.3 • UCR Section 5.2.4.7.1.4.4 • UCR Section 5.2.4.7.1.4.5 • UCR Section 5.2.4.7.1.4.6 • UCR Section 5.2.6.1 • UCR Section 5.2.6.1.1 • UCR Section 5.2.6.1.2 • UCR Section 5.2.6.1.3 • UCR Section 5.2.6.1.4 • UCR Section 5.2.6.2 • UCR Section 5.2.6.3 • UCR Section 5.2.6.4 • UCR Section 5.2.6.5 • UCR Section 5.2.1.5.5 • UCR Section 5.2.1.5.5
E1 CAS (MFR1, DTMF, DP)	No (Europe only)			
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes			
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)			

Table 2. PBX 1 Requirements (continued)

DSN Trunk Interfaces (continued)				
Interface	**Critical**	**Requirements** **Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP)	No	Voice	• MOS (R) • Secure calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • CJCSI 6215.01C
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes			
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	VTC	• ITU-T H.320 (R: PRI only)	• FTR 1080B-2002
DSN Line Interfaces				
2-Wire Analog	Yes	Access	• Directory Number Identification (R) • Analog Line (R) • National ISDN 1/2 Basic Access (R: BRI Only) • Basic Line Test Capabilities (R) • Advanced Line Test Capabilities (C) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (R: 2-WireAnalog only) • Alerting Signals and Tones (R) • S/T Reference Point (R: ISDN BRI only) • VoIP System Requirements (R: VoIP Phones only)	• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.3.5 • UCR Section 5.2.1.3.3 • UCR Section 5.2.1.5.4.1.1 • UCR Section 5.2.1.5.4.1.1 • UCR Section 5.2.4.2.1 • UCR Section 5.2.4.3.1 • UCR Section 5.2.4.5.1 • UCR Section 5.2.4.7.1.2.1 • UCR Section 5.2.12.8
ISDN BRI NI 1/2 (ANSI T1.619a)	No			
2-Wire Proprietary Digital	No	Voice	• MOS (R) • Secure Calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
VoIP (Ethernet IEEE 802.3u)	No	Data	• Modem (VBD) (R: 2-Wire Analog only) • Secure data (STE/STU-III) (R: 2-Wire Analog only)	• CJCSI 6215.01C • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: BRI only)	• FTR 1080B-2002
DSN Features & Capabilities				
Feature/ **Capability**	**Critical**	**Requirements** **Required or Conditional**		**References**
Common Features	Yes	• Individual Lines (R) • Denied originating service (C) • Code restriction and diversion (R) • Call waiting (R) • Three-way calling (R) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (R) • Call Transfer - Internal Only (R) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R) • Call Transfer – Outside (R) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (R) • Conference Calling – Six Way Station Controlled (C) • Call Forwarding Variable (R) • Call Forward Busy Line (R) • Call Forwarding – Don't Answer – All Calls (R) • Selective Call Forwarding (C) • Call pick-up (C) • Address Translation (C) • Assured Dial Tone (R)		• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.1.3 • UCR Section 5.2.1.1.4 • UCR Section 5.2.1.1.5.1 • UCR Section 5.2.1.1.6 • UCR Section 5.2.1.1.7 • UCR Section 5.2.1.1.7.1 • UCR Section 5.2.1.1.7.2 • UCR Section 5.2.1.1.7.3 • UCR Section 5.2.1.1.7.4 • UCR Section 5.2.1.1.7.5 • UCR Section 5.2.1.1.7.6 • UCR Section 5.2.1.1.7.7 • UCR Section 5.2.1.1.7.8 • UCR Section 5.2.1.1.8.1 • UCR Section 5.2.1.1.8.2 • UCR Section 5.2.1.1.8.3 • UCR Section 5.2.1.1.8.4 • UCR Section 5.2.1.1.9.1 • UCR Section 5.2.1.7 • UCR Section 5.2.1.9
Attendant	No	• Attendant Features (C)		• UCR Section 5.2.1.2.2

Table 2. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	Critical	Requirements Required or Conditional	References
Public Safety	Yes	• Emergency Service Basic (911) Caller (R) • Emergency Service (911) Public Safety Answering Service (C) • Enhanced Emergency Service (E911) (C) • Trace of terminating calls (R) • Outgoing call trace (R)	• UCR Section 5.2.1.4.1.1 • UCR Section 5.2.1.4.1.2 • UCR Section 5.2.1.4.1.3 • UCR Section 5.2.1.4.2 • UCR Section 5.2.1.4.3
Conferencing	No	• Preset Conferencing (C) • Meet-Me Conferencing (C) • Progressive Conferencing (C)	• UCR Section 5.2.1.6.1 • UCR Section 5.2.1.6.2 • UCR Section 5.2.1.6.3
Nailed-up Connections	No	• Nailed-Up Connections (C)	• UCR Section 5.2.1.8
DSN Hotline Services	No	• DSN Analog Hotline Service (C)	• UCR Section 5.2.1.12
MLPP	Yes	• MLPP Overview (R) • Preemption in the Network (R) • Network Facility with Lower Precedence Calls (R) • Network Facility with Equal or Higher Precedence Calls (R) • Precedence Call Diversion (R) • Channel Associated Signaling (R) • Primary Rate Interface (R) • Analog Line MLPP (R) • ISDN MLPP Basic Rate Interface (R) • ISDN Primary Rate Interface (R) • Precedence Call Waiting (R) • Call Forwarding (R) • Call Transfer (R) • Call Hold (R) • Three-Way Calling (R) • Call Pickup (C) • Conferencing (C) • Multiline Hunt Group (C) • Community of Interest (C) • MLPP Interaction with EKTS features (C)	• UCR Section 5.2.2.1.1 • UCR Section 5.2.2.2 • UCR Section 5.2.2.2.1 • UCR Section 5.2.2.2.2 • UCR Section 5.2.2.3 • UCR Section 5.2.2.4.1 • UCR Section 5.2.2.4.2 • UCR Section 5.2.2.5 • UCR Section 5.2.2.6 • UCR Section 5.2.2.7 • UCR Section 5.2.2.8.1 • UCR Section 5.2.2.8.2 • UCR Section 5.2.2.8.3 • UCR Section 5.2.2.8.4 • UCR Section 5.2.2.8.5 • UCR Section 5.2.2.8.6 • UCR Section 5.2.2.8.7.1 • UCR Section 5.2.2.8.8 • UCR Section 5.2.2.8.9 • UCR Section 5.2.2.10.1

Table 2. PBX 1 Requirements (continued)

Feature/ Capability	Critical	Requirements Required or Conditional	References
Call Processing	Yes	• Call Treatments (R) • Primary and Alternate Routing (R) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (R) • Termination of Analog Lines (R) • DSN User Dialing (R) • Interswitch and Intraswitch Dialing (R) • Seven-Digit Dialing (R) • Ten-Digit Dialing (R) • Access Code (R) • Access Digit (R) • Precedence Digit (R) • Service Digit (R) • Route Code (R) • Area Code (R) • Switch Code (R) • Line Number (R) • Calling Name Delivery (C) • Calling Number Delivery (R) • Emergency Service 911 Conflict Resolution (R) • DSN Switch Outpulsing Digit Formats (C) • Standard Directory Number (R) • Standard Test Numbers (C) • Base Services – Abbreviated Numbers (R) • Digit Reception Requirements (R) • Screening (R)	• UCR Section 5.2.3.1 • UCR Section 5.2.3.2 • UCR Section 5.2.3.3.1 • UCR Section 5.2.3.3.2 • UCR Section 5.2.3.3.3 • UCR Section 5.2.3.3.4 • UCR Section 5.2.3.5.1.1 • UCR Section 5.2.3.5.1.1 • UCR Section 5.3.3.5.2.1 • UCR Section 5.2.3.5.2.2 • UCR Section 5.2.3.5.1.3 • UCR Section 5.2.3.5.1.3.1 • UCR Section 5.2.3.5.1.3.2 • UCR Section 5.2.3.5.1.3.3 • UCR Section 5.2.3.5.1..4 • UCR Section 5.2.3.5.1.5 • UCR Section 5.2.3.5.1.6 • UCR Section 5.2.3.5.1.7 • UCR Section 5.2.3.5.1.8.1 • UCR Section 5.2.3.5.1.8.2 • UCR Section 5.2.3.5.1.9 • UCR Section 5.2.3.5.2 • UCR Section 5.2.3.5.3 • UCR Section 5.2.3.5.4 • UCR Section 5.2.3.5.5 • UCR Section 5.2.3.5.6 • UCR Section 5.2.3.5.8
ISDN Services	Yes	• BRI Access, Call Control and Signaling (R) • Uniform Interface Configuration for BRIs (R) • EKTS (C) • PRI Access, Call Control and Signaling (R) • PRI Features (R) • Packet Data Features and Capabilities (C)	• UCR Section 5.2.9.2, Table 5.2.9-1 • UCR Section 5.2.9.2, Table 5.2.9-2 • UCR Section 5.2.9.3, Table 5.2.9-3 • UCR Section 5.2.9.2, Table 5.2.9-4 • UCR Section 5.2.9.2, Table 5.2.9-5 • UCR Section 5.2.9.2, Table 5.2.9-6
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 5.2.10.1.1.2 • UCR Section 5.2.10.1.1.2.2 • UCR Section 5.2.10.2 • UCR Section 5.2.10.3 • UCR Section 5.2.10.4
Reliability	Yes	• System Availability (R) • Backup Power (R) • Power Components (R) • UPS Requirements (R) • UPS PBX 1 Load Capacity (R) • Backup Power (Environmental) (R) • Alarms (R)	• UCR Section 5.2.11.2 • UCR Section 5.2.11.3 • UCR Section 5.2.11.3.1 • UCR Section 5.2.11.3.2 • UCR Section 5.2.11.3.2.1 • UCR Section 5.2.11.3.3 • UCR Section 5.2.11.3.4
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Section 3

Table 2. PBX 1 Requirements (continued)

VoIP				
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
VoIP System (See note 1.)	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better (R) • ITU-T G.711 PCM CODEC (R) • MLPP (R) • Security (R) • Network management (C) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging (R) • Softphone Requirements (C)		• UCR section 5.2.12.8.2.1 • UCR section 5.2.12.8.2.2 • UCR section 5.2.12.8.2.3 • UCR section 5.2.12.8.2.4 • UCR section 5.2.12.8.2.5 • UCR section 5.2.12.8.2.6 • UCR section 5.2.12.8.2.7 • UCR 2008, Change 1, section 5.3.5 • UCR section 5.2.12.8.2.9 • UCR 2008, Change 1, section 5.3.2.6.1.7
Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN (See note 2.)	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (R) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.4.2.2 • UCR Section 5.2.4.3.2 • UCR Section 5.2.4.3.4

NOTES:
1   All requirements are derived from the UCR 2008, Reference (e) with the exception of the IPv6 and softphone requirements, because TDM requirements were not included in the UCR 2008, Change 1. The IPv6 and softphone requirements are derived from the UCR 2008, Change 1, Reference (f).
2   Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.

Table 2. PBX 1 Requirements (continued)

LEGEND:					
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	FTR 1080B-2002 G.711	Video Teleconferencing Services PCM of voice frequencies	PCM-24	Pulse Code Modulation - 24 Channels
ANSI	American National Standards Institute	GR GR-815	Generic Requirement Generic Requirements For Network Element/Network	PCM-30	Pulse Code Modulation - 30 Channels
BER	Bit Error Ratio		System (NE/NS) Security	PRI	Primary Rate Interface
BRI	Basic Rate Interface	H.320	Standard for Narrowband VTC	PSTN	Public Switched Telephone Network
C	Conditional	IEEE	Institute of Electrical and Electronics Engineers	Q.955.3	ISDN Signaling Standard for E1 MLPP
CAS	Channel Associated Signaling				
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IP	Internet Protocol	R	Required
		IPv6	Internet Protocol version 6	S/T	ISDN BRI four-wire interface
CODEC	Coder/Decoder	ISDN	Integrated Services Digital Network	SS7	Signaling System 7
DIACAP	DoD Information Assurance Certification and Accreditation Process	IT	Information Technology	STE	Secure Terminal Equipment
		ITU-T	International Telecommunication Union- Telecommunication Standardization Sector	STIGs	Security Technical Implementation Guides
DISA	Defense Information Systems Agency			STU-III	Secure Telephone Unit -3rd generation
DISR	DoD IT Standards Registry				
DoD	Department of Defense	kbps	kilobits per second	T.4	Standardization of Group 3 facsimile terminals for document transmission
DoDI	Department of Defense Instruction	Mbps MFR1	Megabits per second Multi-Frequency Recommendation 1		
DP	Dial Pulse			T1	Digital Transmission Link Level 1 (1.544 Mbps)
DS0	Digital Signal Level 0 (64 kbps)	MLPP	Multi-Level Precedence and Preemption		
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	MOS	Mean Opinion Score	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DSN	Defense Switched Network	NI 1/2	National ISDN Standard 1 or 2	TDM	Time Division Multiplexing
DTMF	Dual Tone Multi-Frequency	NX56	Data format restricted to multiples of 56 kbps	UCR	Unified Capabilities Requirements
E&M	Ear and Mouth				
E1	European Basic Multiplex Rate (2.048 Mbps)	NX64	Data format restricted to multiples of 64 kbps	UPS	Uninterruptible Power Supply
EKTS	Electronic Key Telephone System	PBX	Private Branch Exchange	VBD	Variable bit data
		PBX 1	Private Branch Exchange 1	VoIP	Voice over Internet Protocol
FTR	Federal Telecommunications Recommendation	PCM	Pulse Code Modulation	VTC	Video Teleconferencing

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at https://jit.fhu.disa.mil (NIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: ucco@disa.mil.

JITC Memo, JTE, Special Interoperability Test Certification of Cisco Unified Communications Manager Version 8.0(2) with Internetwork Operating System (IOS) Software Release 15.1(1)T

6. The JITC point of contact is Mr. Edward Mellon, DSN 879-5159, commercial (520) 538-5159, FAX DSN 879-4347, or e-mail to edward.mellon@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 1002901.

FOR THE COMMANDER:

2 Enclosures a/s   for BRADLEY A. CLARK
                                          Chief
                                          Battlespace Communications Portfolio

Distribution (electronic mail):
Joint Staff J-6
Joint Interoperability Test Command, Liaison, TE3/JT1
Office of Chief of Naval Operations, CNO N6F2
Headquarters U.S. Air Force, Office of Warfighting Integration & CIO, AF/XCIN (A6N)
Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ
U.S. Marine Corps MARCORSYSCOM, SIAT, MJI Division I
DOT&E, Net-Centric Systems and Naval Warfare
U.S. Coast Guard, CG-64
Defense Intelligence Agency
National Security Agency, DT
Defense Information Systems Agency, TEMC
Office of Assistant Secretary of Defense (NII)/DOD CIO
U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities
    Division, J68
Defense Information Systems Agency, GS23

ADDITIONAL REFERENCES

(c)   Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Cisco Unified Communications Manager (CUCM) 8.0 (Tracking Number 1002901)," 31 March 2011

(d)   Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)," 9 November 2007

(e)   Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008," 22 January 2009

(f)   Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008 Change 1," 22 January 2010

(g)   Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006

CERTIFICATION TESTING SUMMARY

1. SYSTEM TITLE. Unified Communications Manager Version 8.0(2), with Internetwork Operating System (IOS) Software Release 15.1(1)T; hereinafter referred to as the System Under Test (SUT).

2. PROPONENT. Headquarters United States Air Force Europe (HQ USAFE).

3. PROGRAM MANAGER. Joseph Halcli, HQ USAFE/A6NA, PSC2 Box 11095, APO AE, 09012, e-mail: joseph.halcli@ramstein.af.mil.

4. TESTER. Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

5. SYSTEM UNDER TEST DESCRIPTION. The SUT is a Private Branch Exchange (PBX) 1. The SUT supports American National Standards Institute (ANSI) T1.619a Digital Transmission Link Level 1 (T1) Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) National ISDN Standard 1 or 2 (NI 1/2) and International Telecommunication Union - Telecommunication Standardization Sector ITU-T Q.931 European Basic Multiplex Rate (E1) ISDN PRI trunk interfaces. The SUT consists of Communications Managers running the Cisco Unified Communications Manager software, gateways, and Internet Protocol (IP) telephones. The Cisco Unified Communications Manager is the software-based call-processing component of the Cisco enterprise IP telephone solution. The Cisco Unified Communications Manager software is a client-server application loaded on Cisco 7800 Series Media Convergence Servers (MCSs) or Unified Computing System (UCS) servers with VMware ESXi. The Cisco Communications Manager software provides telephony features and capabilities to packet telephony network devices such as Voice over Internet Protocol (VoIP) phones. The Cisco Unified Communications Managers tested were the MCS 7835-I3, MCS 7835-H2, MCS 7825-I4, UCS C210-M1 with VMware ESXi, and UCS5108 with B200-M1 with VMware ESXi. The other family series of servers which include: the MCS 7835-I2, MCS 7825-H3, MCS 7825-H4, MCS 7835-H3, MCS 7845-H2, MCS 7845-H3, MCS 7845-I2, MCS 7845-I3, and UCS5108 with B250-M1 with VMware ESXi utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.

The 2951, 2851, 3945 and 3845 scalable integrated services routers are included in this tested architecture. The 2951 and 2851 have one Network Module (NM) slot, one High-Density Extension Voice Module (EVM-HD) slot, and four High-Performance Wide Area Network (WAN) Interface Card (WIC) (HWIC) slots. The 2951 can be populated with up to 16 T1 trunks or 52 Foreign Exchange Station (FXS) ports, while the 2851 can be populated with up to 7 T1 trunks or 52 FXS ports. The 2911 and 2921 utilize the same software and similar hardware as the 2951 and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use. The 2811 and 2821 utilize the same software and similar hardware as the 2851 and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use. The 3945 and 3845 have

four NM slots and four HWIC slots. Each NM slot on the 3945 and 3845 can accommodate a standard NM, an enhanced-network-module (NME) or an EVM-HD. The 3945 supports up to 24 T1 trunks or 112 FXS ports. The 3845 supports up to 24 T1 trunks or 88 FXS ports. The 3825, 3945E, 3925, and 3925E utilize the same software and similar hardware as the 3945 and 3845 and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.

Management of the SUT is though a site-provided, Secure Technical Implementation Guide (STIG)-compliant workstation, with Windows Experience (XP) Service Pack (SP)3 installed.

6. OPERATIONAL ARCHITECTURE. The Defense Switched Network (DSN) architecture is a two-level network hierarchy consisting of DSN backbone switches and Service/Agency installation switches. Joint Staff policy and subscriber mission requirements determine which type of switch can be used at a particular location. The DSN architecture, therefore, consists of several categories of switches including PBXs. The Unified Capabilities Requirements (UCR) operational DSN Architecture is depicted in Figure 2-1. The architecture depicts the relationship of Military Department PBX 1s to the other DSN switch types.
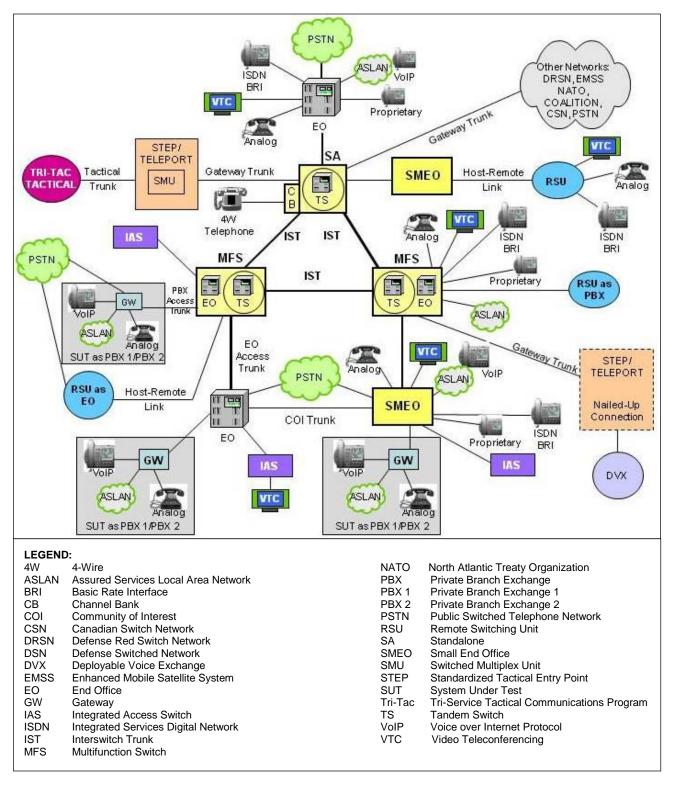
LEGEND:

4W	4-Wire
ASLAN	Assured Services Local Area Network
BRI	Basic Rate Interface
CB	Channel Bank
COI	Community of Interest
CSN	Canadian Switch Network
DRSN	Defense Red Switch Network
DSN	Defense Switched Network
DVX	Deployable Voice Exchange
EMSS	Enhanced Mobile Satellite System
EO	End Office
GW	Gateway
IAS	Integrated Access Switch
ISDN	Integrated Services Digital Network
IST	Interswitch Trunk
MFS	Multifunction Switch

NATO	North Atlantic Treaty Organization
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange 1
PBX 2	Private Branch Exchange 2
PSTN	Public Switched Telephone Network
RSU	Remote Switching Unit
SA	Standalone
SMEO	Small End Office
SMU	Switched Multiplex Unit
STEP	Standardized Tactical Entry Point
SUT	System Under Test
Tri-Tac	Tri-Service Tactical Communications Program
TS	Tandem Switch
VoIP	Voice over Internet Protocol
VTC	Video Teleconferencing

Figure 2-1. DSN Architecture

7. REQUIRED SYSTEM INTERFACES. Requirements specific to PBX 1s are listed in Table 2-1. These requirements are derived from:

    a. DSN services for Network and Applications specified in Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)", Reference (d).

    b. UCR interface and signaling requirements for trunks/lines verified through JITC testing and/or vendor submission of Letters of Compliance (LoC), References (e) and (f).

    c. UCR PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) verified through JITC testing and/or vendor submission of LoC, References (e) and (f).

Table 2-1. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	**Critical**		**Requirements Required or Conditional**	**References**
T1 CAS (MFR1, DTMF, DP)	No	Trunking	• PBX Line (C) • Direct Inward Dialing (C) • National ISDN 1/2 Primary Access (R) • ISDN ANSI MLPP Service Capability (R) • ITU-T ISDN Primary Access (Europe only) (C) • ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C) • Normal Wink Start Operations (R) • Glare Operation (R) • Abnormal Wink Start (R) • Glare Resolution (R) • Call for Service Timing (R) • Guard Timing (R) • Satellite Timing (R) • Disconnect Control (R) • Reselect and Retrial (R) • Off-Hook Supervision Transition (R) • Dial-Pulse Signals (R) • DTMF Signaling (R) • Standard Digit Format for Precedence (C) • MFR1 2/6 Signaling (C) • Alerting Signals and Tones (R) • DSN ISDN User-to-Network Signaling (R) • Application (R) • Physical Layer (R) • Data Link Layer (R) • Data Link Connection (R) • Peer-to-Peer Procedures of Data-Link Layer (R) • Layer 3 DSN User-to-Network Signaling (R) • DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R) • Sequence of Messages for DSN Circuit-Switched Calls (R) • Message Functional Definition and Content (R) • General Message Format and Information Elements Coding (R) • Supplementary Services (C) • PCM-24 Digital Trunk Interface (R) • Interface Characteristics (R) • Supervisory Channel Associated Signaling (R) • Clear Channel Capability (R) • Alarm and Restoral Requirements (R) • PCM-30 Digital Trunk Interface (Europe only) (R) • Interoperation of PCM-24 and PCM-30 (R) • Analog Trunk Interface (C) • Integrated Digital Loop Carrier (C) • Trunk Group-Remove from Service (R) • Trunk Group-Restore to Service (R)	• UCR Section 5.2.1.3.1 • UCR Section 5.2.1.3.2 • UCR Section 5.2.1.3.4.1 • UCR Section 5.2.1.3.4.1.1 • UCR Section 5.2.1.3.4.2 • UCR Section 5.2.1.3.4.2.1 • UCR Section 5.2.4.3.3.1.1 • UCR Section 5.2.4.3.3.1.2 • UCR Section 5.2.4.3.3.2.1 • UCR Section 5..2.4.3.3.2.2 • UCR Section 5.2.4.3.5 • UCR Section 5.2.4.3.6 • UCR Section 5.2.3.4.7 • UCR Section 5.2.3.4.8 • UCR Section 5..2.3.4.9 • UCR Section 5.2.3.4.10 • UCR Section 5..2.4.4.1 • UCR Section 5.2.4.4.2 • UCR Section 5..2.4.4.2.1 • UCR Section 5.2.4.4.3 • UCR Section 5..2.4.5.1 • UCR Section 5.2.4.7.1.4.2 • UCR Section 5..2.4.7.1.1 • UCR Section 5.2.4.7.1.2 • UCR Section 5.2.4.7.1.3 • UCR Section 5..2.4.7.1.3.1 • UCR Section 5.2.4.7.1.3.2 • UCR Section 5.2.4.7.1.4 • UCR Section 5.2.4.7.1.4.2 • UCR Section 5.2.4.7.1.4.3 • UCR Section 5.2.4.7.1.4.4 • UCR Section 5.2.4.7.1.4.5 • UCR Section 5.2.4.7.1.4.6 • UCR Section 5.2.6.1 • UCR Section 5.2.6.1.1 • UCR Section 5.2.6.1.2 • UCR Section 5.2.6.1.3 • UCR Section 5.2.6.1.4 • UCR Section 5.2.6.2 • UCR Section 5.2.6.3 • UCR Section 5.2.6.4 • UCR Section 5.2.6.5 • UCR Section 5.2.1.5.5 • UCR Section 5.2.1.5.5
E1 CAS (MFR1, DTMF, DP)	No (Europe only)			
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes			
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)			

Table 2-1. PBX 1 Requirements (continued)

DSN Trunk Interfaces (continued)				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP)	No	Voice	• MOS (R) • Secure calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Facsimile	• Analog: ITU-T T.4 (R)	• DISR
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • CJCSI 6215.01C
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	VTC	• ITU-T H.320 (R: PRI only)	• FTR 1080B-2002
DSN Line Interfaces				
2-Wire Analog	Yes	Access	• Directory Number Identification (R) • Analog Line (R) • National ISDN 1/2 Basic Access (R: BRI Only) • Basic Line Test Capabilities (R) • Advanced Line Test Capabilities (C) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (R: 2-WireAnalog only) • Alerting Signals and Tones (R) • S/T Reference Point (R: ISDN BRI only) • VoIP System Requirements (R: VoIP Phones only)	• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.3.5 • UCR Section 5.2.1.3.3 • UCR Section 5.2.1.5.4.1.1 • UCR Section 5.2.1.5.4.1.1 • UCR Section 5.2.4.2.1 • UCR Section 5.2.4.3.1 • UCR Section 5.2.4.5.1 • UCR Section 5.2.4.7.1.2.1 • UCR Section 5.2.12.8
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Voice	• MOS (R) • Secure Calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
2-Wire Proprietary Digital	No	Facsimile	• Analog: ITU-T T.4 (R)	• DISR
VoIP (Ethernet IEEE 802.3u)	No	Data	• Modem (VBD) (R: 2-Wire Analog only) • Secure data (STE/STU-III) (R: 2-Wire Analog only)	• CJCSI 6215.01C • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: BRI only)	• FTR 1080B-2002
DSN Features & Capabilities				
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
Common Features	Yes	• Individual Lines (R) • Denied originating service (C) • Code restriction and diversion (R) • Call waiting (R) • Three-way calling (R) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (R) • Call Transfer - Internal Only (R) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R) • Call Transfer – Outside (R) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (R) • Conference Calling – Six Way Station Controlled (C) • Call Forwarding Variable (R) • Call Forward Busy Line (R) • Call Forwarding – Don't Answer – All Calls (R) • Selective Call Forwarding (C) • Call pick-up (C) • Address Translation (C) • Assured Dial Tone (R)		• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.1.3 • UCR Section 5.2.1.1.4 • UCR Section 5.2.1.1.5.1 • UCR Section 5.2.1.1.6 • UCR Section 5.2.1.1.7 • UCR Section 5.2.1.1.7.1 • UCR Section 5.2.1.1.7.2 • UCR Section 5.2.1.1.7.3 • UCR Section 5.2.1.1.7.4 • UCR Section 5.2.1.1.7.5 • UCR Section 5.2.1.1.7.6 • UCR Section 5.2.1.1.7.7 • UCR Section 5.2.1.1.7.8 • UCR Section 5.2.1.1.8.1 • UCR Section 5.2.1.1.8.2 • UCR Section 5.2.1.1.8.3 • UCR Section 5.2.1.1.8.4 • UCR Section 5.2.1.1.9.1 • UCR Section 5.2.1.7 • UCR Section 5.2.1.9
Attendant	No	• Attendant Features (C)		• UCR Section 5.2.1.2.2

Table 2-1. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Public Safety	Yes	• Emergency Service (911) Caller (R) • Emergency Service (911) Public Safety Answering Service (C) • Enhanced Emergency Service (E911) (C) • Trace of terminating calls (R) • Outgoing call trace (R)	• UCR Section 5.2.1.4.1.1 • UCR Section 5.2.1.4.1.2 • UCR Section 5.2.1.4.1.3 • UCR Section 5.2.1.4.2 • UCR Section 5.2.1.4.3
Conferencing	No	• Preset Conferencing (C) • Meet-Me Conferencing (C) • Progressive Conferencing (C)	• UCR Section 5.2.1.6.1 • UCR Section 5.2.1.6.2 • UCR Section 5.2.1.6.3
Nailed-up Connections	No	• Nailed-Up Connections (C)	• UCR Section 5.2.1.8
DSN Hotline Services	No	• DSN Analog Hotline Service (C)	• UCR Section 5.2.1.12
MLPP	Yes	• MLPP Overview (R) • Preemption in the Network (R) • Network Facility with Lower Precedence Calls (R) • Network Facility with Equal or Higher Precedence Calls (R) • Precedence Call Diversion (R) • Channel Associated Signaling (R) • Primary Rate Interface (R) • Analog Line MLPP (R) • ISDN MLPP Basic Rate Interface (R) • ISDN Primary Rate Interface (R) • Precedence Call Waiting (R) • Call Forwarding (R) • Call Transfer (R) • Call Hold (R) • Three-Way Calling (R) • Call Pickup (C) • Conferencing (C) • Multiline Hunt Group (C) • Community of Interest (C) • MLPP Interaction with EKTS features (C)	• UCR Section 5.2.2.1.1 • UCR Section 5.2.2.2 • UCR Section 5.2.2.2.1 • UCR Section 5.2.2.2.2 • UCR Section 5.2.2.3 • UCR Section 5.2.2.4.1 • UCR Section 5.2.2.4.2 • UCR Section 5.2.2.5 • UCR Section 5.2.2.6 • UCR Section 5.2.2.7 • UCR Section 5.2.2.8.1 • UCR Section 5.2.2.8.2 • UCR Section 5.2.2.8.3 • UCR Section 5.2.2.8.4 • UCR Section 5.2.2.8.5 • UCR Section 5.2.2.8.6 • UCR Section 5.2.2.8.7.1 • UCR Section 5.2.2.8.8 • UCR Section 5.2.2.8.9 • UCR Section 5.2.2.10.1

Table 2-1. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Call Processing	Yes	• Call Treatments (R) • Primary and Alternate Routing (R) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (R) • Termination of Analog Lines (R) • DSN User Dialing (R) • Interswitch and Intraswitch Dialing (R) • Seven-Digit Dialing (R) • Ten-Digit Dialing (R) • Access Code (R) • Access Digit (R) • Precedence Digit (R) • Service Digit (R) • Route Code (R) • Area Code (R) • Switch Code (R) • Line Number (R) • Calling Name Delivery (C) • Calling Number Delivery (R) • Emergency Service 911 Conflict Resolution (R) • DSN Switch Outpulsing Digit Formats (C) • Standard Directory Number (R) • Standard Test Numbers (C) • Base Services – Abbreviated Numbers (R) • Digit Reception Requirements (R) • Screening (R)	• UCR Section 5.2.3.1 • UCR Section 5.2.3.2 • UCR Section 5.2.3.3.1 • UCR Section 5.2.3.3.2 • UCR Section 5.2.3.3.3 • UCR Section 5.2.3.3.4 • UCR Section 5.2.3.5.1.1 • UCR Section 5.2.3.5.1.1 • UCR Section 5.3.3.5.2.1 • UCR Section 5.2.3.5.2.2 • UCR Section 5.2.3.5.1.3 • UCR Section 5.2.3.5.1.3.1 • UCR Section 5.2.3.5.1.3.2 • UCR Section 5.2.3.5.1.3.3 • UCR Section 5.2.3.5.1..4 • UCR Section 5.2.3.5.1.5 • UCR Section 5.2.3.5.1.6 • UCR Section 5.2.3.5.1.7 • UCR Section 5.2.3.5.1.8.1 • UCR Section 5.2.3.5.1.8.2 • UCR Section 5.2.3.5.1.9 • UCR Section 5.2.3.5.2 • UCR Section 5.2.3.5.3 • UCR Section 5.2.3.5.4 • UCR Section 5.2.3.5.5 • UCR Section 5.2.3.5.6 • UCR Section 5.2.3.5.8
ISDN Services	Yes	• BRI Access, Call Control and Signaling (R) • Uniform Interface Configuration for BRIs (R) • EKTS (C) • PRI Access, Call Control and Signaling (R) • PRI Features (R) • Packet Data Features and Capabilities (C)	• UCR Section 5.2.9.2, Table 5.2.9-1 • UCR Section 5.2.9.2, Table 5.2.9-2 • UCR Section 5.2.9.3, Table 5.2.9-3 • UCR Section 5.2.9.2, Table 5.2.9-4 • UCR Section 5.2.9.2, Table 5.2.9-5 • UCR Section 5.2.9.2, Table 5.2.9-6
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 5.2.10.1.1.2 • UCR Section 5.2.10.1.1.2.2 • UCR Section 5.2.10.2 • UCR Section 5.2.10.3 • UCR Section 5.2.10.4
Reliability	Yes	• System Availability (R) • Backup Power (R) • Power Components (R) • UPS Requirements (R) • UPS PBX 1 Load Capacity (R) • Backup Power (Environmental) (R) • Alarms (R)	• UCR Section 5.2.11.2 • UCR Section 5.2.11.3 • UCR Section 5.2.11.3.1 • UCR Section 5.2.11.3.2 • UCR Section 5.2.11.3.2.1 • UCR Section 5.2.11.3.3 • UCR Section 5.2.11.3.4
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Section 3

Table 2-1. PBX 1 Requirements (continued)

VoIP			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
VoIP System (See note 1.)	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better (R) • ITU-T G.711 PCM CODEC (R) • MLPP (R) • Security (R) • Network management (C) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging (R) • Softphone Requirements (C)	• UCR section 5.2.12.8.2.1 • UCR section 5.2.12.8.2.2 • UCR section 5.2.12.8.2.3 • UCR section 5.2.12.8.2.4 • UCR section 5.2.12.8.2.5 • UCR section 5.2.12.8.2.6 • UCR section 5.2.12.8.2.7 • UCR 2008, Change 1, section 5.3.5 • UCR section 5.2.12.8.2.9 • UCR 2008, Change 1, section 5.3.2.6.1.7

Network Gateways			
Gateway	**Critical**	**Requirements Required or Conditional**	**References**
PSTN (See note 2.)	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (R) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.4.2.2 • UCR Section 5.2.4.3.2 • UCR Section 5.2.4.3.4

NOTES:
1   All requirements are derived from the UCR 2008, Reference (e) with the exception of the IPv6 and softphone requirements, because TDM requirements were not included in the UCR 2008, Change 1. The IPv6 and softphone requirements are derived from the UCR 2008, Change 1, Reference (f).
2   Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.

Table 2-1. PBX 1 Requirements (continued)

LEGEND:					
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	FTR 1080B-2002	Video Teleconferencing Services	PCM-24	Pulse Code Modulation - 24 Channels
ANSI	American National Standards Institute	G.711	PCM of voice frequencies	PCM-30	Pulse Code Modulation - 30 Channels
BER	Bit Error Ratio	GR	Generic Requirement	PRI	Primary Rate Interface
BRI	Basic Rate Interface	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PSTN	Public Switched Telephone Network
C	Conditional	H.320	Standard for Narrowband VTC	Q.955.3	ISDN Signaling Standard for E1 MLPP
CAS	Channel Associated Signaling	IEEE	Institute of Electrical and Electronics Engineers	R	Required
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IP	Internet Protocol	S/T	ISDN BRI four-wire interface
CODEC	Coder/Decoder	IPv6	Internet Protocol version 6	SS7	Signaling System 7
DIACAP	DoD Information Assurance Certification and Accreditation Process	ISDN	Integrated Services Digital Network	STE	Secure Terminal Equipment
DISA	Defense Information Systems Agency	IT	Information Technology	STIGs	Security Technical Implementation Guides
DISR	DoD IT Standards Registry	ITU-T	International Telecommunication Union-Telecommunication Standardization Sector	STU-III	Secure Telephone Unit - 3rd generation
DoD	Department of Defense			T.4	Standardization of Group 3 facsimile for document transmission
DoDI	Department of Defense Instruction	kbps	kilobits per second		
		Mbps	Megabits per second		
DP	Dial Pulse	MFR1	Multi-Frequency Recommendation 1	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DS0	Digital Signal Level 0 (64 kbps)	MLPP	Multi-Level Precedence and Preemption	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	MOS	Mean Opinion Score	TDM	Time Division Multiplexing
		NI 1/2	National ISDN Standard 1 or 2		
DSN	Defense Switched Network			UCR	Unified Capabilities Requirements
DTMF	Dual Tone Multi-Frequency	NX56	Data format restricted to multiples of 56 kbps		
E&M	Ear and Mouth			UPS	Uninterruptible Power Supply
E1	European Basic Multiplex Rate (2.048 Mbps)	NX64	Data format restricted to multiples of 64 kbps	VBD	Variable bit data
EKTS	Electronic Key Telephone System	PBX	Private Branch Exchange	VoIP	Voice over Internet Protocol
		PBX 1	Private Branch Exchange 1		
FTR	Federal Telecommunications Recommendation	PCM	Pulse Code Modulation	VTC	Video Teleconferencing

8.   TEST NETWORK DESCRIPTION. The SUT was tested at JITC's Global Information Grid Network Test Facility in a manner and configuration similar to that of the DSN operational environment. Testing of the system's required functions and features was conducted using the notional test configuration depicted in Figure 2-2. The SUT test configuration with an Assured Services Local Area Network (ASLAN) is depicted in Figure 2-3. The SUT was tested as the end-point in relation to the other switches.
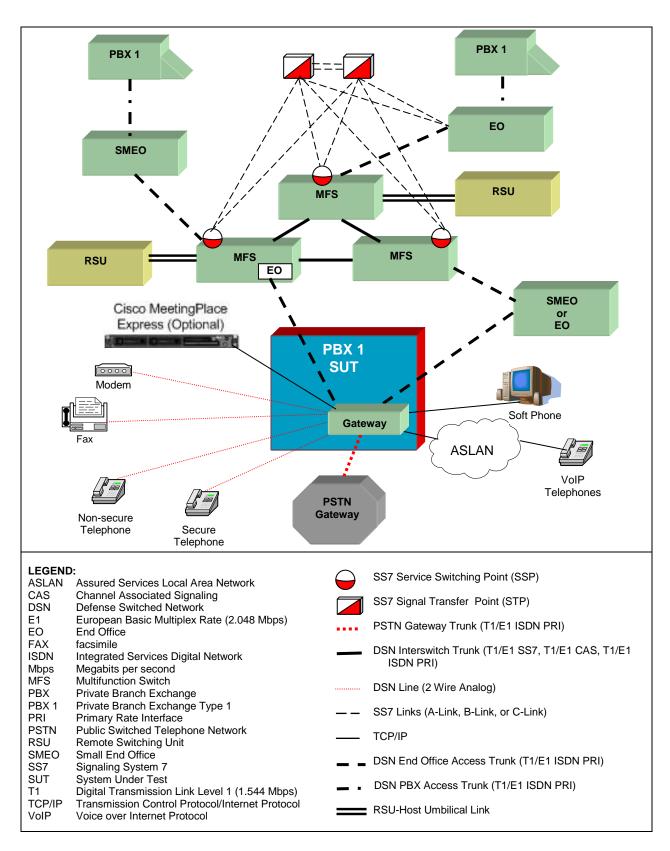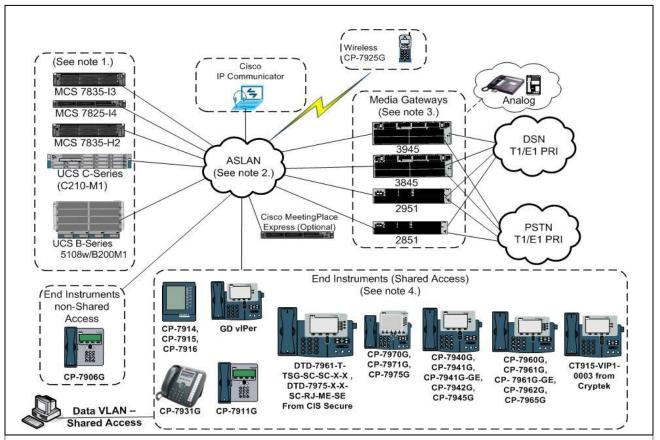
LEGEND:

ASLAN	Assured Services Local Area Network
CAS	Channel Associated Signaling
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
EO	End Office
FAX	facsimile
ISDN	Integrated Services Digital Network
Mbps	Megabits per second
MFS	Multifunction Switch
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange Type 1
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RSU	Remote Switching Unit
SMEO	Small End Office
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TCP/IP	Transmission Control Protocol/Internet Protocol
VoIP	Voice over Internet Protocol

- SS7 Service Switching Point (SSP)
- SS7 Signal Transfer Point (STP)
- PSTN Gateway Trunk (T1/E1 ISDN PRI)
- DSN Interswitch Trunk (T1/E1 SS7, T1/E1 CAS, T1/E1 ISDN PRI)
- DSN Line (2 Wire Analog)
- SS7 Links (A-Link, B-Link, or C-Link)
- TCP/IP
- DSN End Office Access Trunk (T1/E1 ISDN PRI)
- DSN PBX Access Trunk (T1/E1 ISDN PRI)
- RSU-Host Umbilical Link

Figure 2-2. SUT Notional Test Configuration

2-11

NOTES:
1. The MCS 7835-I3, MCS 7835-H2 and MCS 7825-I4 Communications Managers were tested. The other family series of servers which includes the MCS 7835-I2, MCS 7825-H3, MCS 7825-H4, MCS 7835-H3, MCS 7845-H3, MCS 7845-H2, MCS 7845-I2, and MCS 7845-I3 utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.
2. The SUT is certified with any ASLAN or combination of certified ASLAN components listed on the Unified Capabilities Approved Products List.
3. The 2951, 2851, 3945, and 3845 Integrated Service Routers were tested. The other family series, which includes the 2811, 2121, 3825, 3945E, 3925, and 3925E utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.
4. Refer to paragraph 11.a.(5)(a)10.c. for certified shared access rates.
5. All components enclosed in dashed lines are the SUT.

LEGEND:

ASLAN	Assured Services Local Area Network	MCS	Media Convergence Server
CP	Cisco Phone	PRI	Primary Rate Interface
DSN	Defense Switched Network	PSTN	Public Switched Telephone Network
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test
GD	General Dynamics	T1	Digital Transmission Link Level 1 (1.544 Mbps)
IP	Internet Protocol	UCS	Unified Computing System
JITC	Joint Interoperability Test Command	VLAN	Virtual Local Area Network
Mbps	Megabits per second		

Figure 2-3. SUT Test Configuration with ASLAN

9. SYSTEM CONFIGURATIONS. Table 2-2 provides the system configurations, hardware and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine interoperability with a complement of DSN switches noted in Table 2-2. Table 2-2 lists the DSN switches which depict the tested configuration and is not intended to identify the only switches that are certified with the SUT. The SUT is certified with switching systems listed on the Unified Capabilities (UC) Approved Products List (APL) that offer the same certified interfaces.

Table 2-2. Tested System Configurations

System Name	Software Release		
Avaya CS2100	Succession Enterprise (SE) 09.1		
Siemens EWSD	19d with Patch Set 46		
Redcom HDX	V.3.0a (R3P0)		
Avaya S8720	Communication Manager (CM) 4.0 (R014x.00.2.732.1: Super Patch 16538)		
Cisco MeetingPlace® Express	2.1		
Cisco Unified Communications Manager Version 8.0(2), with IOS Software Release 15.1(1)T			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Description**
Communications Managers MCS 7835-I3, **MCS 7835-H2**, **MCS 7825-I4,** MCS 7835-I2, MCS 7825-H3, MCS 7825-H4, MCS 7835-H3, MCS 7845-H3, MCS 7845-H2, MCS 7845-I3, MCS 7845-I2	8.0.2.40003-3	Not Applicable	Processing/Signaling
UCS C Series Server UCS C210-M1 with VMware ESXi	8.0.2.40003-3	Not Applicable	Processing/Signaling
UCS Server UCS5108 with B200-M1 and B250-M1 with VMware ESXi	8.0.2.40003-3	**6120XP** with version ucs-manager-k9.1.2.1b	BIOS
		Cisco MDS 9124-FC Switch with version 4.1(3a)	Fiber Channel Switch
		EMC AX4 SAN: NaviSphere Express AX4-5 with version 02.23.050.5.703	Hard Drive
Cisco 3845, 3825 Integrated Service Router **Cisco 3945**, 3945E, 3925, 3925E Integrated Service Router Generation 2 (Gateway**)**	IOS 15.1(1)T	**NM HDV2**	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 2.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 2.)
		VIC3 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VIC3 2FXS	Voice Interface Card, 2-port, Foreign Exchange Station
		SM-NM-ADAPTER	Available only in 2951/3945 for SM (Service Module) to NM (Network Module) adaption
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 3.)
		EVM3 HDA 8FXS/DID	8-Port HD analog and digital extension module for voice and fax (See note 3.)
		PVDM2	Digital Signal Processor (See note 4.)

Table 2-2. Tested System Configurations (continued)

Cisco Unified Communications Manager Version 8.0(2), with IOS Software Release 15.1(1)T (continued)			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Description**
Cisco 2951, 2921, 2911, **2851**, 2821, 2811 Integrated Services Router (Gateway)	IOS 15.1(1)T	**NM HD V2**	2-slot IP communications enhanced voice/fax network module
		SM-NM-ADAPTER	Available only in 2951/3945 for SM (Service Module) to NM (Network Module) adaption
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		EVM3 HD 8FXS/DID	HD analog and digital extension module for voice and fax
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 3.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 2.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 3.)
		VIC3 4FXS/DID	Voice interface card, 4-port, RJ-11, foreign exchange station, DID
		VIC3 2FXS	Voice Interface card, 2-port, RJ-11, Foreign exchange station
		PVDM2	Digital Signal Processor (See note 4.)
CP-7940G and CP-7960G (See note 5.)	P00308010200	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7970G and CP-7971G	SCCP70.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7931G	SCCP31.9-0-2SR1S	Not Applicable	IP Phone (with push to talk handset or with standard handset)
CP-7911G and 7906G	SCCP11.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE	SCCP41.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7942G and CP-7962G	SCCP42.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7945G and CP-7965G	SCCP45.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7975G	SCCP75.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
7914	Load: S00105000400	Not Applicable	Expansion module
7915	B015-1-0-4	Not Applicable	Expansion module
7916	B015-1-0-4	Not Applicable	Expansion module
General Dynamics C4 Systems Sectéra® vIPer™ (See note 6.)	Release 1.0, Software ver.6.04	Not Applicable	IP Phone (with standard handset)
CIS Secure DTD-7961-T-SG-SC-SC-X-X (See note 7.)	SCCP41.9-0-2SR1S	Not Applicable	7961G TEMPEST version with 100 Mbps SC Fiber LAN and PC interfaces, TSG Positive Disconnect, no speakerphone, shared access

Table 2-2. Tested System Configurations (continued)

Cisco Unified Communications Manager Version 8.0(2), with IOS Software Release 15.1(1)T (continued)			
Component **(See note 1.)**	**Release**	**Sub-component** **(See note 1.)**	**Description**
CIS Secure DTD-7975-X-XSC-RJ-ME-SE (See note 7.)	SCCP75.9-0-2SR1S	Not Applicable	7975G Standard with 1000 Mbps SC Fiber LAN and RJ45 PC interfaces, shared access
CRYPTEK CT915-V-P1-003 (See note 7.)	SCCP41.9-0-2SR1S	Not Applicable	7961G IP phone, Fiber TEMPEST version with 100MB Fiber LAN and no shared access
Walker WS-2620	Not Applicable	Not Applicable	Push to Talk Handset for Cisco 7900 Series phones
Cisco IP Communicator (See note 8.)	7.0.3	Not Applicable	site-provided STIG-compliant PC with Windows XP SP 3
Management Workstation	Not Applicable	Not Applicable	site-provided STIG-compliant PC with Windows XP SP 3
CP-7925G	CP7925G-1.3.3	Not Applicable	Wireless IP phone
The following phones were tested with the SUT, but are not certified for use with the SUT due to critical failures.			
Telecore 2151	2AE-00056-0102	Not Applicable	IP Phone (with push-to-talk handset or with standard handset), 100 Mbps shared access[9]
L-3 Communications IP STE	1.2.4	Not Applicable	IP STE[11]

NOTES:
1   Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.
2   These components are certified in the DISN with T1 ISDN PRI ANSI T1 619a and E1 T1 PRI (ITU-T Q.955.3) interfaces. These components are certified in the PSTN with the T1 ISDN PRI (ANSI T1.607) and E1 ISDN PRI ITU-T Q.931) interfaces.
3   The EM HDA 8FXS and EM3 HDA 8FXS/DID expansion modules require the EVM HD module. Up to two EM HDA 8FXS or EM3 HDA 8FXS/DID expansion modules are supported for each EVM HD.
4   The 2800, 2900, 3800, and 3900 series of Integrated Service Routers (Gateways) are certified with the Packet Voice Digital Signal Processor Module 2 (PVDM2). Initial testing of the Packet Voice Digital Signal Processor Module 3 (PVDM3) in the 2951 and 3945 Gateways showed excessive one-way latency. Testing on all gateways was completed with the PVDM2. The PVDM3 is not certified for use with the SUT.
5   The SUT met all IPv6 requirements through testing and LoC with the following exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with Reference (g). This was adjudicated by DISA as having a minor operational impact.
6   This instrument is certified specifically with 2800 and 3800 series gateways with IOS 12.4(22) T2 or higher version listed on the UC APL.
7   CIS Secure Computing and Cryptek add security modifications which are physical in nature and do not affect software. The CIS Secure Computing and Cryptek security modifications can be applied to any certified Cisco IP phone.
8   The Cisco IP Communicator was tested on the Windows XP and Windows Vista operating systems platforms. The Cisco IP Communicator with Windows Vista not certified due to inability to do DSCP tagging. The Cisco IP Communicator is only certified on the Windows XP operating system platform.
9   The Telecore 2151 was tested with the SUT; however, it is not certified with the SUT due excessive loss of voice media. The Telecore 2151 failed to meet the Packet Voice Impairment Test (PVIT) requirements when tested with other phones.
10 Calls could not be placed from the L-3 IP STE when it was tested with the SUT. Although the L-3 IP STE certified with previous CUCM versions, it is not certified with this version of the CUCM.

Table 2-2. Tested System Configurations (continued)

LEGEND:

ANSI	American National Standards Institute	HDA	High Density Analog	Q.955.3	ISDN Signaling Standard for E1 MLPP
APL	Approved Product List	HDX	High Density Exchange	RJ	Registered Jack
BIOS	Basic Input/Output System	IOS	Internetwork Operating System	SC	fiber connector (square push-in)
CP	Cisco Phone	IP	Internet Protocol	SCCP	Skinny Call Control Protocol
CS	Communication Server	Ipv4	Internet Protocol version 4	SM	Service Module
DID	Direct Inward Dialing	IPv6	Internet Protocol version 6	SP3	Service Pack 3
DISN	Defense Information System Network	ISDN	Integrated Services Digital Network	SS7	Signaling System 7
DSCP	Differentiated Services Code Point	ITU-T	International Telecommunication Union- Telecommunication Standardization Sector	STIG	Secure Technical Implementation Guide
DSS1	Digital Subscriber Signaling 1	JITC	Joint Interoperability Test Command	SUT	System Under Test
E1	European Basic Multiplex Rate (2.048 Mbps)	LAN	Local Area Network	T1	Digital Transmission Link Level 1 (1.544 Mbps)
EM	Expansion Module	LoC	Letters of Compliance	T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
EVM	Extension Voice Module	Mbps	Megabits per second		
EWSD	Elektronisches Wählsystem Digital	MCS	Media Convergence Server	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
		MFT	Multiflex Trunk		
Fax	facsimile	MLPP	Multi-Level Precedence and Preemption	TDM	Time Division Multiplexing
FXS	Foreign Exchange Station	NM	Network Module	UC	Unified Capabilities
G	10/100BaseT Ethernet (A Cisco part designator on their IP phone.)	PC	Personal Computer	UCR	Unified Capabilities Requirements
		PRI	Primary Rate Interface		
		PSTN	Public Switched Telephone Network	UCS	Unified Computing System
GE	Gigabit Ethernet (A Cisco part designator on their IP phone.)			V	Voice
		PVDM2	Packet Voice Digital Signal Processor Module 2	VIC	Voice Interface Card
				VWIC	Voice WAN Interface Card
HD	High Density	Q.931	Signaling Standard for ISDN	WAN	Wide Area Network

10. TESTING LIMITATIONS. None.

11. TEST RESULTS

a. Discussion

(1) DSN Trunk Interfaces

(a) The SUT met all critical CRs and FRs for the T1 ISDN PRI NI 1/2 (ANSI T1.619a) and E1 ISDN PRI (ITU-T Q.955.3) interfaces with one minor exception. The SUT does not support Non Facility Associated Signaling (NFAS) on their T1 ISDN PRI NI2 interface in accordance with the UCR. DISA previously adjudicated this anomaly as having a minor operational impact and stated the intent to change this from required to conditional for a PBX 1. The T1 Channel Associated Signaling (CAS) interface is supported by the SUT; however, the T1 CAS interface is not certified due to the following critical discrepancies: The SUT T1 CAS interface does not recognize Remove from Service (Busy Out) or Restore to Service (Make Idle) condition from the distant end switch.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for 2-Wire Loop Start Analog (GR-506-CORE), 2-Wire Ground Start Analog (GR-506-CORE), and Voice over Internet Protocol (VoIP) DSN line interfaces with the minor exceptions listed in paragraphs 11.a.(3)(a) and 11.a.(5)(a)8.

(3) Features and Capabilities

(a) Common Features. The SUT met all critical interoperability certification requirements for Features and Capabilities with the following exceptions: All of the features on the VoIP phones were tested using multiple line appearances. Although the SUT does not support standard or precedence Call Waiting on their VoIP phones, they do support multiple call appearances, which mitigates this discrepancy. This provides the ability for a user to receive additional calls while active with another call. A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. These discrepancies were adjudicated by DISA as minor on 25 August 2010.

(b) Attendant. This feature is not supported by the SUT and is not required for a PBX 1.

(c) Public Safety. The SUT meets the minimum critical interoperability requirements for Public Safety which is basic emergency service 911 service. This feature allows the user to dial 911 and the SUT then retranslates it to be routed to a Public Safety Answering Point via a trunk or line. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. These public safety features are not required for a PBX 1.

(d) Conferencing. The SUT does not support conferencing. However, the SUT can support Meet-Me Conferencing through the use of an optional adjunct conferencing system called the Cisco MeetingPlace® Express, which is covered under a separate certification. The SUT does not support Preset Conferencing or Progressive Conferencing. These features are not required for a PBX 1.

(e) Nailed-up Connections. This feature is not supported by the SUT and is not required for a PBX 1.

(f) Multi-Level Precedence and Preemption (MLPP). Met all critical CRs and FRs with the following minor exceptions:

1. The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified Communications Manager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.

2. The SUT does not support the Loss of Command and Control (C2) announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. This anomaly was previously adjudicated as minor because this announcement would rarely be invoked on a PBX 1.

(g) Call Processing. Met all critical CRs and FRs.

(h) ISDN Services. Met all critical CRs and FRs for the T1 PRI interface with the following minor exception: The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA previously adjudicated this anomaly as having a minor operational impact and stated the intent to change this from required to conditional for a PBX 1.

(i) Synchronization. All critical interoperability certification CRs and FRs were met for this feature by the SUT. The SUT supports line timing mode and Internal Stratum 4 for synchronization.

(j) Reliability. All critical interoperability certification CRs and FRs for this feature were met by the SUT and met by vendor LoC.

(l) Security. Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (c).

(4) Network Gateways. The SUT met all critical interoperability certification requirements for the Public Switched Telephone Network (PSTN) Network Gateway. The interfaces certified for the PSTN are T1 ISDN PRI NI 1/2 (ANSI T1.607), E1 ISDN PRI (ITU-T Q.931), and 2-Wire Analog Ground Start Line (GR-506 CORE). The SUT offers a T1 CAS trunk interface; however, it was not certified. Critical interoperability discrepancies (refer to paragraph 11.a(1)a) were discovered during testing. The SUT T1 CAS interface is not certified for use within the DISN. This is not a required interface for a PBX 1.

(5) VoIP. The SUT is certified with any ASLAN or any combination of certified ASLAN components listed on the UC APL.

(a) VoIP System. The UCR, section 5.2.12.8.2, outlines the requirements for the VoIP system. The VoIP system requirements encompass end-to-end VoIP requirements. The following paragraphs detail the results of the SUT VoIP solution.

1. Voice Quality. In accordance with the UCR, section 5.2.12.8.2.1, VoIP calls shall have an average Mean Opinion Score (MOS) of at least 4.0 as measured in accordance with ITU-T P.800 voice quality standards. This applies from handset to handset and for intra- and inter-switch calls end-to-end. The SUT meets MOS requirements with an average of 4.33 for 110 test calls. The SUT met this requirement with all VoIP phones to include the Cisco IP Communicator softphone.

2. Codec. In accordance with the UCR, section 5.2.12.8.2.2, the ITU-T G.711 Pulse Code Modulation (PCM) CODEC with a 20 milliseconds (ms) packet fill was required and was met by the SUT VoIP solution.

3. MLPP. In accordance with the UCR, section 5.2.12.8.2.3, the VoIP system shall meet all MLPP requirements identified in UCR, section 3. All critical MLPP features and functions were met.

4. Security. Security requirements in accordance with the UCR, section 5.2.12.8.2.4, are verified using the Information Assurance Test Plan. Results of the security testing are reported in a separate test report generated by the DISA Information Assurance test personnel, Reference (c).

5. Network Management (NM). In accordance with the UCR, section 5.2.12.8.2.5, the vendor is required to provide a management system to monitor the performance of the ASLAN portion of the VoIP system. This requirement was covered under a separate certification for the respective ASLANs listed on the UC APL. In accordance with the UCR, section 5.3.8, the switching system NM requirements are not required for a PBX 1 and were not tested.

6. Synchronization. In accordance with the UCR, section 5.2.10.1.1.2, the SUT is required to derive timing with line timing mode and an internal clock of stratum 4 or better. The SUT met this requirement.

7. Latency. The UCR, section 5.2.12.8.2.7, states that one-way system latency for the VoIP system must be 60 ms or less as averaged over any five-minute period. The latency requirement is measured from IP or analog handset to the egress trunk. The SUT one way latency measurements were conducted from each phone type supported by the SUT for IPv4 and IPv6 traffic. There were twelve, 20-minute inter switch wired IP phone calls with a measured latency between 54.3 ms to 58.9 ms, with an average of 56.93 ms. In addition there were two 1 hour and 15 minute inter switch wireless IP phone calls placed from the CP-7925G with a measured average one-way latency between 63 ms to 69 ms with an average of 66.59 ms.

8. Internet Protocol version 6 (IPv6). In accordance with UCR, section 5.3.5, all systems submitted for testing must be IPv6 capable. Dual Stack solutions are preferred and tunneling solutions are unacceptable. IPv6-capable products, in accordance with UCR, section 5.3.5.3.1, can create or receive, process, and send or forward IPv6 packets in mixed IPv4/v6 environments. IPv6-capable networks can receive, process, and forward IPv6 packets from/to devices within the same network and from/to other networks and systems, where those networks and systems may be operating with only IPv4, only IPv6, or both IPv4 and IPv6. All of the SUT components covered under this certification met the IPv6 criteria through testing and the LoC with the following minor exceptions, which were adjudicated by DISA as having a minor operational impact:

a. The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other end instruments listed in Table 2-2 and a

dual stack call control agent in accordance with Reference (g). This was adjudicated by DISA as having a minor operational impact.

b. During initial boot up of the CP-7940G and CP-7960G phones, some of the User Datagram Protocol (UDP)/Trivial File Transfer Protocol (TFTP) traffic has a Differentiated Services Code Point (DSCP) value of 4 and 802.1Q value of 5 and cannot be changed.

c. The SUT management workstation provided during testing did not assign DSCP values for Operational Administration and Maintenance (OAM) IP traffic.

d. The IP phones are incorrectly tagging IPv6 Transmission Control Protocol (TCP) traffic during power up.

e. The soft Client is incorrectly tagging all traffic during power up.

f. The 802.1Q COS tag values are not independently configurable from the DSCP values.

g. End Instruments do not support the manual configuration of the IPv6 default gateway.

h. Communications Managers are incorrectly tagging UDP/TFTP traffic to the end instrument. IPv6 Traffic Class and IPv4 DSCP values for signaling cannot be set to the full range of 0-63 in accordance with the UCR. The SUT can only tag Traffic Class and DSCP values for signaling with the following values: 0, 8, 10, 12, 14, 16, 18, 20, 22, 24, 26, 28, 30, 32, 34, 36, 40, 46, 48, and 56.

9. In accordance with the UCR, section 5.2.12.8.2.9, the VoIP system (i.e. Media Gateway and Session Control Agent) shall meet the following requirements:

a. All components shall be capable of implementing Service Class tagging using the 8-bit Traffic Class in the IPv6 header and DSCP field in the IPv4 header. The SUT meets the requirement.

b. All session control components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Table 5.3.1-3. The SUT session control components can only have the signaling service class traffic configured for 21 different DSCP values and not the full range required. The Traffic Class and DSCP values for media can be assigned to any value from 0-63. The MCS 7835 and the MCS 7825 OAM traffic is tagged at zero and is not configurable. In addition, the 2851 and 3845 gateways are tagging IPv4 RTP Control Protocol (RTCP) traffic at zero and it is not configurable. These discrepancies were adjudicated by DISA as having a minor operational impact.

c. For VoIP, video, and data end products, any end system that supports convergence must preassign the VLAN using Institute of Electrical and Electronics Engineers (IEEE) 802.1Q tags prior to the frames entering the ASLAN in accordance with UCR, section 5.3.1.7.4. For end-systems that support just one media, the LAN can assign the VLAN based on port-based VLAN assignment. The SUT Communications Manager does not support more than one media; therefore, VLAN tagging is not supported. There is no operational impact.

10. In accordance with the UCR, section 5.2.12.8.2.9, the VoIP system end user devices shall meet the following requirements:

a. All end instruments shall be capable of implementing Service Class tagging using the 8-bit Traffic Class in the IPv6 header and DSCP field in the IPv4 header. The SUT end instruments that support IPv6 dual stack used class tagging in the respective IP headers for IPv4 and IPv6, which meets the requirement.

b. All end instrument components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Table 5.3.1-3. The DSCPs may be assigned by either having the end instrument itself assign the Traffic Class and DSCP tag to the distinct service class or having the call control portion of the VoIP system tell the end instrument what distinct service class to assign. The SUT end instruments components only have the ability to configure 21 different DSCP values for signaling service class traffic. The DSCP values for media can be assigned to any DSCP value from 0-63. The DSCP value of traffic on the CP-7940 and CP-7960 phones is configured to 4, and it cannot be changed. These discrepancies were adjudicated by DISA as having a minor operational impact. A management workstation that meets the requirement for assigning DSCP values in the IPv4 header was not provided for test.

c. For VoIP, video, and data end products, any end system that supports convergence must preassign the VLAN using IEEE 802.1Q tags prior to the frames entering the ASLAN in accordance with UCR, section 5.3.1.7.4. For end-systems that support just one media, the LAN can assign the VLAN based on port-based VLAN assignment. The SUT end instruments have the capability of supporting shared access. Additionally the SUT end instruments have the capability to tag Real Time Traffic with the appropriate VLAN Identifier value. The Cisco VoIP phones that met the critical interoperability requirements for certification with 100 Mbps interface were the: CP-7906G, CP-7911G, CP-7925G (wireless), CP-7931, CP-7940G, CP-7941G, CP-7941G-GE, CP-7942G, CP-7945G, CP-7960G, CP-7961G-GE, CP-7961G, CP-7962G, CP-7965G, CP-7970G, CP-7971G-GE, CP-7975G, Tempest phone Cryptek 7961G, Tempest phone CIS 7961G, and Tempest phone CIS 7975G. The above phones have been tested and are certified for shared access (i.e., same switch port is shared by PC and IP phone) with the exception of the CP-7906G and CP-7925G (wireless). The CP-7906G and CP-7925G (wireless) phones do not support shared access. The following phones are also certified for 1 Gbps shared access: CP-7971G-GE, CP-7975G, CP-7965G, CP-7945G, CP-7941G-GE, CP-7961G-GE, and Tempest phone CIS 7975G. The CP-7970G and CP-7971G-GE phones are capable of web

browsing; however, this feature was not tested, is not covered by this certification. All VoIP phones were tested using Secure Real Time Protocol (SRTP) which encrypts the media stream. The SRTP is able to encrypt only IP phone to IP phone intra-switch traffic and IP phone to gateway intra-switch traffic. All other voice calls (i.e. analog to analog, or analog to gateway traffic) are not encrypted.

        11.   The UCR 2008, Change 1, section 5.3.2.6.1.7, states that the softphone shall be conceptually identical to a traditional IP "hard" telephone and is required to provide voice features and functionality provided by a traditional IP hard telephone, unless explicitly stated here within this paragraph. The softphone application in conjunction with a general-purpose computer, including its mouse (point and click) interaction, shall support, as a minimum, the UCR 2008, Change 1, section 5.3.2 requirements. The softphone is exempt from the Network Infrastructure End-to-End requirements in the UCR 2008, Change 1, section 5.3.3. The softphone shall meet the Information Assurance requirements in the UCR 2008, Change 1, section 5.4. Information Assurance is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (c).

The Cisco IP Communicator was tested on the Windows XP and Windows Vista operating systems platforms. The Cisco IP Communicator with Windows Vista not certified due to inability to do DSCP tagging. The Cisco IP Communicator is only certified on the Windows XP operating system platform. The Cisco IP Communicator software is loaded on a site-provided Secure Technical Implementation Guide (STIG)-compliant PC with Windows XP SP 3 with the following minimum requirements: Pentium P4 1.5 GHz or higher recommended, 100 MB free disk space, 1 GB RAM, a non-ISA full-duplex sound card (integrated or PCI-based) or USB sound Device, a 10/100 Mbps Ethernet network interface card, SVGA video card 800 x 600 x16-bit screen resolution (1024 x 768 x 16-bit or better recommended). If Cisco VPN Client software is installed, version 5.0 or later is required. If Cisco AnyConnect is installed, version 2.5 is required. For connectivity, a 128 kbps or faster network connection is recommended and adding Cisco Unified Video Advantage with a connection of 384 kbps or faster is required.

    b. System Interoperability Results. The SUT is certified for joint use in the Defense Information System Network (DISN) as a PBX 1 and PBX 2 in accordance with the requirements set forth in References (e) and (f). The identified test discrepancies that remained open after software patches were applied and regression testing was completed have an overall minor operational impact. The SUT interoperability test summary is shown in Table 2-3. The SUT Interoperability Requirements/Status is shown in Table 2-4.

Table 2-3. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	The SUT T1 CAS interface did not meet all critical CRs and FRs. The SUT T1 CAS interface is not certified by JITC and is not required for a PBX 1.[1]
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	Although this interface is offered by the SUT, it was not tested. The SUT E1 CAS interface is not certified by JITC and is not required for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI2 interface does not support NFAS.[2]
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Met all critical CRs and FRs.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog Loop Start (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT gateway analog interface does not provide a ping ring when the phone is configured with the Call Forward Variable feature.[3]
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 1.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 1.
VoIP (Ethernet IEEE 802.3u)	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The Cisco IP phones do not support Call Waiting.[4] The Cisco CP-7940G and CP-7960G are legacy end instruments that did not meet dual stack IPv6 requirements.[5]
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exception: Full compliance of DSN Common Call Features was not met.[3]
Attendant	No	Not Tested	This feature is not supported by the SUT and is not required for a PBX 1.
Public Safety	Yes	Certified	The SUT met all critical CRs and FRs for Basic 911.[6]
Conferencing	No	Not Tested (See note 7.)	The SUT does not support conferencing. However, the SUT can support Meet-Me Conferencing with the optional Cisco MeetingPlace® Express. The SUT does not support Preset Conferencing or Progressive Conferencing and these features are not required for a PBX 1.
Nailed-up Connections	No	Not Tested	This feature is not supported by the SUT and is not required for a PBX 1.
DSN Hotline Services	No	Not Tested	This feature is not supported by the SUT and is not required for a PBX 1.
MLPP	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support a global diversion number.[8] The SUT does not support the Loss of Command and Control announcement.[9]
Call Processing	Yes	Certified	Met all critical CRs and FRs.
ISDN Services	Yes	Certified	Met all critical CRs and FRs on the T1 PRI interface with the following minor exception: The SUT T1 ISDN PRI NI2 interface does not support NFAS.[2]
Synchronization	Yes	Certified	Met all critical CRs and FRs.
Reliability	Yes	Certified	Met all critical CRs and FRs.
Security	Yes	Certified	See note 10.
VoIP System	No	Certified	The SUT is certified for VoIP with any certified ASLAN or ASLAN components posted on the UC APL. The SUT also met IPv4/IPv6 requirements. (See notes 5 and 11.)
VoIP Softphone	No	Certified	Met all critical CRs and FRs.

Table 2-3. SUT Interoperability Test Summary (continued)

Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	The SUT T1 CAS interface did not meet all critical CRs and FRs. The SUT T1 CAS interface is not certified by JITC and is not required for a PBX 1.[1]
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	Although this interface is offered by the SUT, it was not tested. The SUT E1 CAS interface is not certified by JITC and is not required for a PBX 1.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI/2 interface does not support NFAS.[2]
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	2-Wire Analog Ground Start (GR-506-CORE)	No	Certified	Met all critical CRs and FRs.[12]

NOTES:
1   The SUT T1 CAS interface does not recognize Remove from Service (Busy Out) or Restore to Service (Make Idle) condition from the distant end switch. These are critical requirements for this interface; therefore, this interface is not certified by JITC.
2   The SUT does not support NFAS on their T1 ISDN PRI NI2. DISA previously adjudicated this anomaly as having a minor operational impact and stated the intent to change this from required to conditional for a PBX 1.
3   A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.
4   All of the features on the VoIP phones were tested using multiple line appearances. Although the SUT does not support Precedence Call Waiting on their VoIP phones, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call.
5   The SUT met all IPv4 and IPv6 requirements through testing and LoC with the following exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in Enclosure 2 and a dual stack call control agent in accordance with Reference (g). This was adjudicated by DISA as having a minor operational impact.
6   The SUT only supports emergency basic 911 service. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. These public safety features are not required for a PBX 1.
7   The SUT does not support conferencing. However, the SUT can support Meet-Me Conferencing through the use of an optional adjunct conferencing system called the Cisco MeetingPlace® Express, which is covered under a separate certification.
8   The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified Communications Manager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.
9   The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. DISA previously adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1.
10   Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (c).
11   The SUT met all IPv4 and IPv6 requirements with the following discrepancies noted with the SUT, which were adjudicated by DISA as having a minor operational impact:
    a. The VoIP SUT session control components and end instruments can only have the signaling service Traffic Class configured for 21 different DSCP values and not the full range required of 0-63.
    b. The MCS 7835 and the MCS 7825 call managers OAM traffic is tagged at zero and is not configurable.
    c. The 2851 and 3845 gateways are tagging IPv4 RTCP traffic at zero and it is not configurable.
    d. When the CP-7940G and CP-7960G phones are powered up, some of the UDP/TFTP traffic has a DSCP value of 4 and 802.1Q value of 5 and cannot be changed.
    e. The SUT management workstation provided during testing did not assign DSCP values for OAM IP traffic.
    f. The IP phones are incorrectly tagging IPv6 TCP traffic during power up.
    g. The Soft Client is incorrectly tagging all traffic during power up.
    h. The 802.1Q CoS tag values are not independently configurable from the DSCP values.
    i. End Instruments do not support the manual configuration of the IPv6 default gateway.
    j. Communications Managers are incorrectly tagging UDP/TFTP traffic to the end instrument during end instrument power up.
12   This interface requirement was met by the vendor's LoC.

Table 2-3. SUT Interoperability Test Summary (continued)

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LEGEND:
802.1Q    Standards for Local and Metropolitan Area      LoC        Letters of Compliance
          Networks: Virtual Bridged Local Area Networks  LSSGR      Local Access and Transport Area (LATA) Switching
802.3u    Standard for carrier sense multiple access with           Systems Generic Requirements
          collision detection at 100 Mbps                Mbps       Megabits per second
ANSI      American National Standards Institute           MCS        Media Convergence Servers
APL       Approved Products List                          MFR1       Multi-Frequency Recommendation 1
ASLAN     Assured Services Local Area Network             MLPP       Multi-Level Precedence and Preemption
BRI       Basic Rate Interface                            NI 1/2     National ISDN Standard 1 or 2
C2        Command and Control                             NI2        National ISDN Standard 2
CAS       Channel Associated Signaling                    NFAS       Non Facility Associated Signaling
CoS       Class of Service                                OAM        Operational Administration and Maintenance
CP        Cisco Phone                                     PBX 1      Private Branch Exchange 1
CRs       Capability Requirements                         PRI        Primary Rate Interface
DISA      Defense Information Systems Agency              PSTN       Public Switched Telephone Network
DP        Dial Pulse                                      Q.931      Signaling Standard for ISDN
DSCP      Differentiated Services Code Point              Q.955.3    ISDN Signaling standard for E1 MLPP
DSN       Defense Switched Network                        RTCP       RTP Control Protocol
DSS1      Digital Subscriber Signaling 1                  RTP        Real-time Transport Protocol
DTMF      Dual Tone Multi-Frequency                       SS7        Signaling System 7
E1        European Basic Multiplex Rate (2.048 Mbps)      SUT        System Under Test
FRs       Feature Requirements                            T1         Digital Transmission Link Level 1 (1.544 Mbps)
GR        Generic Requirement                             T1.607     ISDN Layer 3 Signaling Specification for Circuit Switched
GR-506    LSSGR: Signaling for Analog Interfaces                     Bearer Service for DSS1
IEEE      Institute of Electrical and Electronics Engineers T1.619a  SS7 and ISDN MLPP Signaling Standard for T1
IP        Internet Protocol                               TCP        Transmission Control Protocol
IPv4      Internet Protocol version 4                     TFTP       Trivial File Transfer Protocol
IPv6      Internet Protocol version 6                     UC         Unified Capabilities
ISDN      Integrated Services Digital Network             UCR        Unified Capabilities Requirements
ITU-T     International Telecommunication Union -          UDP        User Datagram Protocol
          Telecommunication Standardization Sector        VoIP       Voice over Internet Protocol
JITC      Joint Interoperability Test Command
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12. TEST AND ANALYSIS REPORT. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: ucco@disa.mil.

Table 2-4. SUT Interoperability Requirements/Status

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 CAS (MFR1, DTMF, DP)	No	Not Certified (See note 1.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Met	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Not Met	See note 1.
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Not Met	See note 1.
				Normal Wink Start Operations (C)	UCR Section 5.2.4.3.3.1.1	Met	
				Glare Operation (C)	UCR Section 5.2.4.3.3.1.2	Met	
				Abnormal Wink Start (C)	UCR Section 5.2.4.3.3.2.1	Met	
				Glare Resolution (C)	UCR Section 5.2.4.3.3.2.2	Met	
				Call for Service Timing (R)	UCR Section 5.2.4.3.5	Met	
				Guard Timing (R)	UCR Section 5.2.4.3.6	Met	
				Satellite Timing (C)	UCR Section 5.2.4.3.7	Met	
				Disconnect Control (C)	UCR Section 5.2.4.3.8	Met	
				Reselect and Retrial (C)	UCR Section 5.2.4.3.9	Not Tested	See note 2.
				Off-Hook Supervision Transition (C)	UCR Section 5.2.4.3.10	Met	
				Dial-Pulse Signals (C)	UCR Section 5.2.4.4.1	Met	
				DTMF Signaling (C)	UCR Section 5.2.4.4.2	Met	
				Standard Digit Format for Precedence (C)	UCR Section 5.2.4.4.2.1	Met	
				MFR1 2/6 Signaling (C)	UCR Section 5.2.4.4.3	Met	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Met	
				DSN Transmission Interface (R)	UCR Section 5.2.5	Met	
				PCM-24 Digital Trunk Interface (R)	UCR Section 5.2.6.1	Met	
				Interface Characteristics (R)	UCR Section 5.2.6.1.1	Met	
				Supervisory Channel Associated Signaling (C)	UCR Section 5.2.6.1.2	Met	
				Clear Channel Capability (R)	UCR Section 5.2.6.1.3	Met	
				Alarm and Restoral Requirements (R)	UCR Section 5.2.6.1.4	Met	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested	See note 2.
				Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Met	
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Not Tested (See note 3.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.1	Not Tested	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Not Tested	
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Not Tested	
				Normal Wink Start Operations (C)	UCR Section 5.2.4.3.3.1.1	Not Tested	
				Glare Operation (C)	UCR Section 5.2.4.3.3.1.2	Not Tested	
				Abnormal Wink Start (C)	UCR Section 5.2.4.3.3.2.1	Not Tested	
				Glare Resolution (C)	UCR Section 5.2.4.3.3.2.2	Not Tested	
				Call for Service Timing (R)	UCR Section 5.2.4.3.5	Not Tested	
				Guard Timing (R)	UCR Section 5.2.4.3.6	Not Tested	
				Satellite Timing (C)	UCR Section 5.2.4.3.7	Not Tested	
				Disconnect Control (C)	UCR Section 5.2.4.3.8	Not Tested	
				Reselect and Retrial (C)	UCR Section 5.2.4.3.9	Not Tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.4.3.10	Not Tested	
				Dial-Pulse Signals (C)	UCR Section 5.2.4.4.1	Not Tested	
				DTMF Signaling (C)	UCR Section 5.2.4.4.2	Not Tested	
				Standard Digit Format for Precedence (C)	UCR Section 5.2.4.4.2.1	Not Tested	
				MFR1 2/6 Signaling (C)	UCR Section 5.2.4.4.3	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Not Tested	
				DSN Transmission Interface (R)	UCR Section 5.2.5	Not Tested	
				PCM-30 Digital Trunk Interface (C)	UCR Section 5.2.6.2	Not Tested	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested	
				Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested	

Table 2-4. SUT Interoperability Requirements/Status (continued)

Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
DSN Trunk Interfaces							
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Met	
				National ISDN 1/2 Primary Access (R)	UCR Section 5.2.1.3.4.1	Partially Met	See note 4.
				ISDN ANSI MLPP Service Capability (R)	UCR Section 5.2.1.3.4.1.1	Met	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Met	
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Met	
				Call for Service Timing (R)	UCR Section 5.2.4.3.5	Met	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Met	
				DSN ISDN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4.2	Met	
				Application (R)	UCR Section 5.2.4.7.1.1	Met	
				Physical Layer (R)	UCR Section 5.2.4.7.1.2	Met	
				Data Link Layer (R)	UCR Section 5.2.4.7.1.3	Met	
				Data Link Connection (R)	UCR Section 5.2.4.7.1.3.1	Met	
				Peer-to-Peer Procedures of Data-Link Layer (R)	UCR Section 5.2.4.7.1.3.2	Met	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4	Met	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.2.4.7.1.4.2	Met	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.2.4.7.1.4.3	Met	
				Message Functional Definition and Content (R)	UCR Section 5.2.4.7.1.4.4	Met	
				General Message Format and Information Elements Coding (R)	UCR Section 5.2.4.7.1.4.5	Met	
				Supplementary Services (C)	UCR Section 5.2.4.7.1.4.6	Not Tested	See note 2.
				DSN Transmission Interface (R)	UCR Section 5.2.5	Met	
				PCM-24 Digital Trunk Interface (R)	UCR Section 5.2.6.1	Met	
				Interface Characteristics (R)	UCR Section 5.2.6.1.1	Met	
				Clear Channel Capability (R)	UCR Section 5.2.6.1.3	Met	
				Alarm and Restoral Requirements (R)	UCR Section 5.2.6.1.4	Met	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Met	
				Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Met	
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				64 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				NX56 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				NX64 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
			VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Not Tested	See note 2.

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Table 2-4. SUT Interoperability Requirements/Status (continued)

Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
			DSN Trunk Interfaces				
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Met	
				ITU-T ISDN Primary Access (C)	UCR Section 5.2.1.3.4.2	Met	
				ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (C)	UCR Section 5.2.1.3.4.2.1	Met	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Not Tested	See note 2.
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Not Tested	See note 2.
				Call for Service Timing (R)	UCR Section 5.2.4.3.5	Met	
				Disconnect Control (C)	UCR Section 5.2.3.4.8	Met	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.3.4.10	Met	
				DSN ISDN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4.2	Met	
				Application (R)	UCR Section 5.2.4.7.1.1	Met	
				Physical Layer (R)	UCR Section 5.2.4.7.1.2	Met	
				Data Link Layer (R)	UCR Section 5.2.4.7.1.3	Met	
				Data Link Connection (R)	UCR Section 5.2.4.7.1.3.1	Met	
				Peer-to-Peer Procedures of Data-Link Layer (R)	UCR Section 5.2.4.7.1.3.2	Met	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4	Met	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.2.4.7.1.4.2	Met	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.2.4.7.1.4.3	Met	
				Message Functional Definition and Content (R)	UCR Section 5.2.4.7.1.4.4	Met	
				General Message Format and Information Elements Coding (R)	UCR Section 5.2.4.7.1.4.5	Met	
				PCM-30 Digital Trunk Interface (C)	UCR Section 5.2.6.2	Met	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested	See note 2.
				Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested	See note 2.
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				64 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				NX56 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				NX64 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	See note 2.
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
			VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Not Tested	See note 2.

Table 2-4. SUT Interoperability Requirements/Status (continued)

DSN Line Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
2-Wire Loop Start Analog	Yes	Certified	Access	Directory Number Identification (R)	UCR Section 5.2.1.1.1	Met	
				PBX Line (C)	UCR Section 5.2.1.3.1	Met	
				Analog Line (R)	UCR Section 5.2.1.3.5	Met	
				Basic Line Test Capabilities (R)	UCR Section 5.2.1.5.4.1.1	Met	
				Advanced Line Test Capabilities (C)	UCR Section 5.2.1.5.4.1.1	Not Tested	See note 2.
				Loop Start Line (R: 2-Wire Analog only)	UCR Section 5.2.4.2.1	Met	
				Reverse Battery (R)	UCR Section 5.2.4.3.1	Met	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Met	
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested (See note 4.)	Access	Directory Number Identification (R)	UCR Section 5.2.1.1.1	Not Tested	
				National ISDN 1/2 Basic Access (C)	UCR Section 5.2.1.3.3	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Not Tested	
				S/T Reference Point (R)	UCR Section 5.2.4.7.1.2.1	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested	
			VTC	ITU-T H.320 (R: BRI only)	FTR 1080B-2002	Not Tested	
2-Wire Proprietary Digital	No	Not Tested (See note 4.)	Access	Directory Number Identification (R)	UCR Section 5.2.1.1.1	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	

Table 2-4. SUT Interoperability Requirements/Status (continued)

Feature/ Capability	Critical	Feature Status	UCR Requirement	Reference	Test Results	Remarks
DSN Features and Capabilities						
Common Features	Yes	Certified	Individual Lines (R)	UCR Section 5.2.1.1.1	Met	
			Denied originating service (C)	UCR Section 5.2.1.1.3	Not Tested	See note 2.
			Code restriction and diversion (C)	UCR Section 5.2.1.1.4	Met	
			Call waiting (R)	UCR Section 5.2.1.1.5.1	Partially Met	See note 6.
			Three-way calling (R)	UCR Section 5.2.1.1.6	Met	See note 6.
			Add-on transfer, conference calling, and call hold (C)	UCR Section 5.2.1.1.7	Met	See note 6.
			Call Transfer Individual – All calls (R)	UCR Section 5.2.1.1.7.1	Met	See note 6.
			Call Transfer - Internal Only (R)	UCR Section 5.2.1.1.7.2	Met	See note 6.
			Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R)	UCR Section 5.2.1.1.7.3	Met	See note 6.
			Call Transfer – Outside (R)	UCR Section 5.2.1.1.7.4	Met	See note 6.
			Call Transfer – Add-On Restricted Station (C)	UCR Section 5.2.1.1.7.5	Not Tested	See note 2.
			Call Transfer – Attendant (C)	UCR Section 5.2.1.1.7.6	Not Tested	See note 2.
			Call Hold (R)	UCR Section 5.2.1.1.7.7	Met	See note 6.
			Conference Calling – Six Way Station Controlled (C)	UCR Section 5.2.1.1.7.8	Partially Met	See note 6.
			Call Forwarding Variable (R)	UCR Section 5.2.1.1.8.1	Partially Met	See note 6.
			Call Forward Busy Line (R)	UCR Section 5.2.1.1.8.2	Met	See note 6.
			Call Forwarding – Don't Answer – All Calls (R)	UCR Section 5.2.1.1.8.3	Met	See note 6.
			Selective Call Forwarding (C)	UCR Section 5.2.1.1.8.4	Met	
			Call pick-up (C)	UCR Section 5.2.1.1.9.1	Met	See note 6.
			Address Translation (C)	UCR Section 5.2.1.7	Met	
			Assured Dial Tone (C)	UCR Section 5.2.1.9	Met	
Attendant	No	Not Tested	Attendant Features (C)	UCR Section 5.2.1.2.2	Not Tested	See note 2.
Public Safety	Yes	Certified	Emergency Service Basic (911) Caller (R)	UCR Section 5.2.1.4.1.1	Met	See note 7.
			Emergency Service (911) Public Safety Answering Service (C)	UCR Section 5.2.1.4.1.2	Not Tested	See note 7.
			Enhanced Emergency Service (E911) (C)	UCR Section 5.2.1.4.1.3	Not Tested	See note 7.
			Trace of terminating calls (C)	UCR Section 5.2.1.4.2	Not Tested	See note 7.
			Outgoing call trace (C)	UCR Section 5.2.1.4.3	Not Tested	See note 7.

Table 2-4. SUT Interoperability Requirements/Status (continued)

Feature/ Capability	Critical	Feature Status	UCR Requirement	Reference	Test Results	Remarks
DSN Features and Capabilities						
Conferencing	No	Not Tested	Preset Conferencing (C)	UCR Section 5.2.1.6	Not Tested	See note 2.
			Meet-Me Conferencing (R)	UCR Section 5.2.1.6.2	Not Tested	See note 8.
			Progressive Conferencing (C)	UCR Section 5.2.1.6.3	Not Tested	See note 2.
Nailed-up Connections	No	Not Tested	Nailed-Up Connections (C)	UCR Section 5.2.1.8	Not Tested	See note 2.
DSN Hotline Services	No	Certified	DSN Analog Hotline Service (C)	UCR Section 5.2.1.12	Met	
MLPP	Yes	Certified	MLPP Overview (R)	UCR Section 5.2.2.1.1	Met	
			Preemption in the Network (R)	UCR Section 5.2.2.2	Met	
			Network Facility with Lower Precedence Calls (R)	UCR Section 5.2.2.2.1	Met	
			Network Facility with Equal or Higher Precedence Calls (R)	UCR Section 5.2.2.2.2	Met	
			Precedence Call Diversion (R)	UCR Section 5.2.2.3	Met	See note 9.
			Channel Associated Signaling (C)	UCR Section 5.2.2.4.1	Met	See note 1.
			Primary Rate Interface (R)	UCR Section 5.2.2.4.2	Met	
			Analog Line MLPP (R)	UCR Section 5.2.2.5	Met	
			ISDN MLPP Basic Rate Interface (C)	UCR Section 5.2.2.6	Not Tested	See note 4.
			ISDN Primary Rate Interface (R)	UCR Section 5.2.2.7	Met	
			Precedence Call Waiting (R)	UCR Section 5.2.2.8.1	Met	See note 5.
			Call Forwarding (R)	UCR Section 5.2.2.8.2	Partially Met	See note 5.
			Call Transfer (R)	UCR Section 5.2.2.8.3	Met	See note 5.
			Call Hold (R)	UCR Section 5.2.2.8.4	Met	See note 5.
			Three-Way Calling (R)	UCR Section 5.2.2.8.5	Met	See note 5.
			Call Pickup (C)	UCR Section 5.2.2.8.6	Met	See note 5.
			Conferencing (C)	UCR Section 5.2.2.8.7.1	Met	See note 8.
			Multiline Hunt Group (C)	UCR Section 5.2.2.8.8	Met	
			Community of Interest (C)	UCR Section 5.2.2.8.9	Not Tested	See note 2.
			MLPP Interaction with EKTS features (C)	UCR Section 5.2.2.10.1	Not Tested	See note 2.

Table 2-4. SUT Interoperability Requirements/Status (continued)

Feature/ Capability	Critical	Feature Status	UCR Requirement	Reference	Test Results	Remarks
DSN Features and Capabilities						
Call Processing	Yes	Certified	Call Treatments (R)	UCR Section 5.2.3.1	Met	
			Primary and Alternate Routing (C)	UCR Section 5.2.3.2	Met	
			E&M Lead Signaling States (C)	UCR Section 5.2.3.3.1	Not Tested	See note 5.
			4-Wire Analog User Access Lines (C)	UCR Section 5.2.3.3.2	Not Tested	See note 5.
			2-Wire User Access Lines (R)	UCR Section 5.2.3.3.3	Met	
			Termination of Analog Lines (R)	UCR Section 5.2.3.3.4	Met	
			DSN User Dialing (R)	UCR Section 5.2.3.5.1.1	Met	
			Interswitch and Intraswitch Dialing (R)	UCR Section 5.2.3.5.1.1	Met	
			Seven-Digit Dialing (R)	UCR Section 5.3.3.5.2.1	Met	
			Ten-Digit Dialing (R)	UCR Section 5.2.3.5.2.2	Met	
			Access Code (R)	UCR Section 5.2.3.5.1.3	Met	
			Access Digit (R)	UCR Section 5.2.3.5.1.3.1	Met	
			Precedence Digit (R)	UCR Section 5.2.3.5.1.3.2	Met	
			Service Digit (R)	UCR Section 5.2.3.5.1.3.3	Met	
			Route Code (R)	UCR Section 5.2.3.5.1.4	Met	
			Area Code (R)	UCR Section 5.2.3.5.1.5	Met	
			Switch Code (R)	UCR Section 5.2.3.5.1.6	Met	
			Line Number (R)	UCR Section 5.2.3.5.1.7	Met	
			Calling Name Delivery (C)	UCR Section 5.2.3.5.1.8.1	Not Tested	See note 2.
			Calling Number Delivery (R)	UCR Section 5.2.3.5.1.8.2	Met	
			Emergency Service 911 Conflict Resolution (R)	UCR Section 5.2.3.5.1.9	Met	
			DSN Switch Outpulsing Digit Formats (C)	UCR Section 5.2.3.5.2	Met	See note 1.
			Standard Directory Number (R)	UCR Section 5.2.3.5.3	Met	
			Standard Test Numbers (C)	UCR Section 5.2.3.5.4	Not Tested	See note 2.
			Base Services – Abbreviated Numbers (C)	UCR Section 5.2.3.5.5	Not Tested	See note 2.
			Digit Reception Requirements (R)	UCR Section 5.2.3.5.6	Met	
			Screening (C)	UCR Section 5.2.3.5.8	Met	
ISDN Services	Yes	Certified	BRI Access, Call Control and Signaling (C)	UCR Section 5.2.9.2, Table 5.2.9-1	Not Tested	See note 2.
			Uniform Interface Configuration for BRIs (C)	UCR Section 5.2.9.2, Table 5.2.9-2	Not Tested	See note 2.
			Electronic Key Telephone Systems (EKTS) (C)	UCR Section 5.2.9.2, Table 5.2.9-3	Not Tested	See note 2.
			PRI Access, Call Control and Signaling (R)	UCR Section 5.2.9.2, Table 5.2.9-4	Partially Met	See note 4.
			PRI Features (R)	UCR Section 5.2.9.2, Table 5.2.9-5	Partially Met	See note 4.
			Packet Data Features and Capabilities (C)	UCR Section 5.2.9.2, Table 5.2.9-6	Not Tested	See note 2.

Table 2-4. SUT Interoperability Requirements/Status (continued)

Feature/Capability	Critical	Feature Status	UCR Requirement	Reference	Test Results	Remarks
DSN Features and Capabilities						
Synchroniz-ation	Yes	Certified	Line timing mode (R)	UCR Section 5.2.11.2	Met	
			Internal Stratum 4 (R)	UCR Section 5.2.10.1.1.2.2	Met	
			Synchronization Performance Monitoring Criteria (C)	UCR Section 5.2.10.2	Not Tested	See note 2.
			DS1 Traffic Interfaces (C)	UCR Section 5.2.10.3	Not Tested	See note 2.
			DS0 Traffic Interconnects (C)	UCR Section 5.2.10.4	Not Tested	See note 2.
Reliability	Yes	Certified	System Availability (R)	UCR Section 5.2.11.2	Met	
			Backup Power (R)	UCR Section 5.2.11.3	Not Tested	See note 10.
			Power Components (R)	UCR Section 5.2.11.3.1	Not Tested	See note 10.
			UPS Requirements (R)	UCR Section 5.2.11.3.2	Not Tested	See note 10.
			UPS PBX 1 Load Capacity (R)	UCR Section 5.2.11.3.2.1	Not Tested	See note 10.
			Backup Power (Environmental) (R)	UCR Section 5.2.11.3.3	Not Tested	See note 10.
			Alarms (R)	UCR Section 5.2.11.3.4	Not Tested	See note 10.
Security	Yes	Certified	GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	UCR Section 3	Met	See note 11.
VoIP						
Feature/Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
VoIP System	No	Certified (See note 12.)	Voice Quality with MOS of 4.0 or better (R)	UCR Section 5.2.12.8.2.1	Met	
			ITU-T G.711 PCM CODEC (R)	UCR Section 5.2.12.8.2.2	Met	
			MLPP (R)	UCR Section 5.2.12.8.2.3	Met	
			Security (R)	UCR Section 5.2.12.8.2.4	Met	
			Network management (C)	UCR Section 5.2.12.8.2.5	Met	
			System timing (R)	UCR Section 5.2.12.8.2.6	Met	
			Latency ≤ 60 milliseconds (R)	UCR Section 5.2.12.8.2.7	Met	
			IPv6 capable (R)	UCR Section 5.2.12.8.2.8	Partially Met	See notes 13 and 14 f, i, j.
			Service Class Tagging (R)	UCR Section 5.2.12.8.2.9	Met	See notes 14 a, b, c, d, e, f, g, h, k.
			Softphone Requirements (C)	UCR 2008, Change 1, Section 5.3.2.6.1.7	Met	

Table 2-4. SUT Interoperability Requirements/Status (continued)

Network Gateways							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
PSTN (See note 15.)	No	Certified	Trunking	Positive Identification Control (C)	CJCSI 6215.01C	Met	
				On-Netting (C)	CJCSI 6215.01C	Met	
				Off-Netting (C)	CJCSI 6215.01C	Met	
				Ground Start Line (R)	UCR Section 5.2.2	Met	See note 16.
				Immediate Start (C)	UCR Section 5.3.2	Met	
				Delay Dial (C)	UCR Section 5.3.4	Met	

NOTES:
1   The SUT T1 CAS interface does not recognize Remove from Service (Busy Out) or Restore to Service (Make Idle) condition from the distant end switch. These are critical requirements for this interface; therefore, this interface is not certified by JITC.
2   This feature/capability is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
3   Although this interface is offered by the SUT, it was not tested. The SUT E1 CAS interface is not certified by JITC and is not required for a PBX 1.
4   The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA previously adjudicated this anomaly as having a minor operational impact and stated the intent to change this from required to conditional for a PBX 1.
5   This interface is not supported by the SUT and is not required for a PBX 1.
6   A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact. All of the features on the VoIP phones were tested using multiple line appearances. Although the SUT does not support Precedence Call Waiting on their VoIP phones, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call.
7   The SUT only supports emergency service 911 public safety features. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. These public safety features are not required for a PBX 1.
8   The SUT does not support conferencing. However, the SUT can support Meet-Me Conferencing through the use of an optional adjunct conferencing system called the Cisco MeetingPlace® Express, which is covered under a separate certification.
9   The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified Communications Manager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.
10   This requirement is a non-testable requirement. It is the responsibility of the respective base/post/camp/station communications agency to provide this with the SUT when installed.
11   Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (c).
12   The SUT is certified for VoIP with any certified ASLAN or ASLAN components posted on the UC APL.
13   The SUT met all IPv4 and IPv6 requirements through testing and LoC with the following exception : The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with Reference (g).   This was adjudicated by DISA as having a minor operational impact.

Table 2-4. SUT Interoperability Requirements/Status (continued)

NOTES (continued):

14 The SUT met all IPv4 and IPv6 requirements with the following discrepancies noted with the SUT, which were adjudicated by DISA as having a minor operational impact:

    a. The VoIP SUT session control components and end instruments can only have the signaling service Traffic Class configured for 21 different DSCP values and not the full range required for 0-63.

    b. The MCS 7835 and the MCS 7825 call managers OAM traffic is tagged at zero and is not configurable.

    c. The 2851 and 3845 gateways are tagging IPv4 RTCP traffic at zero and it is not configurable.

    d. When the CP-7940G and CP-7960G phones are powered up, some of the UDP/TFTP traffic has a DSCP value of 4 and 802.1Q value of 5 and cannot be changed.

    e. The SUT management workstation provided during testing did not assign DSCP values for OAM IP traffic.

    f. The IP phones are incorrectly tagging IPv6 TCP traffic during power up.

    g. The Soft Client is incorrectly tagging all traffic during power up.

    h. The 802.1Q CoS tag values are not independently configurable from the DSCP values.

    i. End Instruments do not support the manual configuration of the IPv6 default gateway.

    j. Communications Managers are incorrectly tagging UDP/TFTP traffic to the end instrument during end instrument power up.

15 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.

16 This interface requirement was met by the vendor's letter of compliance.

LEGEND:

ANSI	American National Standards Institute	FTR 1080B	Video Teleconferencing Services	PCM-24	Pulse Code Modulation - 24 Channels
APL	Approved Products List	-2002		PCM-30	Pulse Code Modulation - 30 Channels
ASLAN	Assured Services Local Area Network	G.711	PCM of voice frequencies	PRI	Primary Rate Interface
BER	Bit Error Ratio	GR	Generic Requirement	PSTN	Public Switched Telephone Network
BRI	Basic Rate Interface	GR-815	Generic Requirements For Network	Q.955.3	ISDN Signaling Standard for E1 MLPP
C	Conditional		Element/Network System (NE/NS) Security	R	Required
CAS	Channel Associated Signaling	H.320	Standard for Narrowband VTC	RTCP	RTP Control Protocol
CJCSI	Chairman of the Joint Chiefs of Staff	IP	Internet Protocol	RTP	Real-time Transport Protocol
	Instruction	IPv4	Internet Protocol version 4	S/T	ISDN BRI 4-wire interface
CODEC	Coder/Decoder	IPv6	Internet Protocol version 6	SS7	Signaling System 7
CP	Cisco Phone	ISDN	Integrated Services Digital Network	STE	Secure Terminal Equipment
DIACAP	DoD Information Assurance Certification and	IT	Information Technology	STIGs	Security Technical Implementation Guides
	Accreditation Process	ITU-T	International Telecommunication Union -	STU-III	Secure Telephone Unit -3rd generation
DISA	Defense Information Systems Agency		Telecommunication Standardization Sector	SUT	System Under Test
DISR	DoD IT Standards Registry	kbps	kilobits per second	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DoD	Department of Defense	Mbps	Megabits per second	T1.619a	SS7 and ISDN MLPP Signaling Standard for
DoDI	Department of Defense Instruction	MCS	Media Convergence Server		T1
DP	Dial Pulse	MFR1	Multi-Frequency Recommendation 1	T.4	Standardization of Group 3 facsimile terminals
DS0	Digital Signal Level 0 (64 kbps)	MLPP	Multi-Level Precedence and Preemption		for document transmission
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048	MOS	Mean Opinion Score	TCP	Transmission Control Protocol
	Mbps European)	NFAS	Non Facility Associated Signaling	TFTP	Trivial File Transfer Protocol
DSCP	Differentiated Services Code Point	NI 1/2	National ISDN Standard 1 or 2	UC	Unified Capabilities
DSN	Defense Switched Network	NI2	National ISDN Standard 2	UCR	Unified Capabilities Requirements
DTMF	Dual Tone Multi-Frequency	NX56	Data format restricted to multiples of 56 kbps	UDP	User Datagram Protocol
E&M	Ear and Mouth	NX64	Data format restricted to multiples of 64 kbps	UPS	Uninterruptible Power Supply
E1	European Basic Multiplex Rate (2.048 Mbps)	OAM	Operational Administration and Maintenance	VBD	Variable bit data
EKTS	Electronic Key Telephone System	PBX	Private Branch Exchange	VoIP	Voice over Internet Protocol
FTR	Federal Telecommunications	PBX 1	Private Branch Exchange 1	VTC	Video Teleconferencing
	Recommendation	PCM	Pulse Code Modulation		